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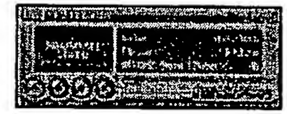

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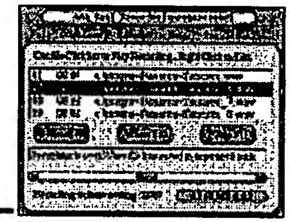
AAA Real Recorder is a sound recording program. It records sound generated, or requested, by other computer programs, such as RealPlayer, Windows Media Player, Quick Time,WinAmp, and many others. The resulting files are saved in wav-file , MP3-file format.



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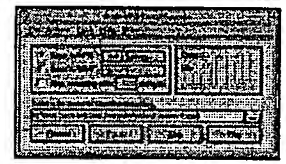
Audio MP3 Sound Recorder offers CD-quality recordings to MP3/WAV. Support recording time length. Recording source is from sound card that means any sound through your sound card would be recorded. Cool skins could be changed to what you like. MP3 VBR is supported with lame encoder for good outputs.



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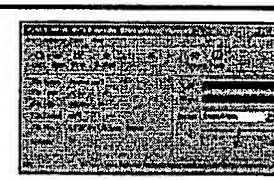
ACE-HIGH MP3 Recorder is professional software designed for recording audio from your computer to hard disk file of MP3 or WAV formats. Comparing with other recording products, our software supports recording to MP3 directly without temporary WAV files generated. With this feature, you may save your hard disk space. And when the ACE-HIGH MP3 Recorder recording, the saved MP3/WAV file's size is calculated and showed on the control panel at real time.



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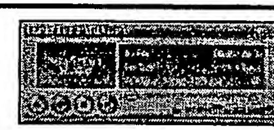
Audio Record Wizard is a sound recorder software, which offers professional recording features with mp3 support.
The audio recorder program is designed to work directly with your sound card, so it can record almost all audio from your sound card at near-perfect quality. So, you can record sound from a microphone, line-in, and just about any other programs (like winamp, realplay, windows media player). audio record wizard can also record



directly to mp3 format if you choose, saving you valuable disk space.

Register It (US$24.95) Download Free Trial

Audio MP3/WMA Recorder makes a complete sound recorder studio of your computer. Sound quality of the recordings remains excellent, or sounds even better for it reduces noise. Audio MP3/WMA Recorder is able to automatically detect the recording formats your sound card supports and then set the application's parameters for the best possible performance. The recordings may be saved as Mp3,Wma or Wav files.

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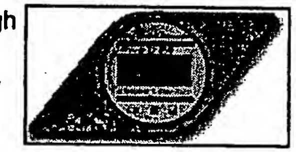
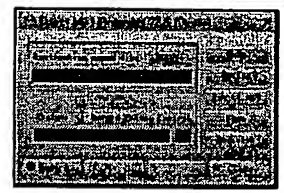


All Sound Recorder enables you to record sound, played back through your sound card and any other sound sources like microphone , VCR, Audio tape player etc. You can use it to grab any sound, including music, dialogs from movies, game sounds etc. from your local computer or the internet. Captured sounds can be saved in WAV or MP3 format, using real-time conversion (without creating temporary files).

Register It (US$19.95) Download Free Trial



Super MP3 Recorder records your computer's audio streaming into MP3 or WAV audio formats. It supports recording sound source from microphone, Internet audio streaming, Winamp, Windows Media Player, Quick Time, Real Player, Flash, Games, etc.

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"FAQ Premium Home Answer" eVoice,
http://content.evoice.com/wcs/signUp/FAQ_premHA_s01.htm

eVoice

FAQ

Premium Home Answer

s It For Me?

FAQ

Demo

Sign Me Up!

What is eVoice?

- What does eVoice do?
- What else do I get with eVoice?
- Where is eVoice available?
- Can eVoice answer my wireless phone?

How does eVoice work?

Is eVoice compatible with other features and equipment on my phone?

What other features does eVoice offer?

What is the privacy policy of eVoice?

How do I subscribe, and what happens after I sign up?

Have Customer Service Questions?

What is eVoice?

What does eVoice do?
eVoice is similar to phone company voicemail, but far better. eVoice offers you more f either the phone company or an answering machine. We'll deliver your voicemail direct email inbox and to your home phone. We'll take calls for you when your phone is busy, online and when you're away from home. eVoice will answer after 4-6 rings, or after as as you choose - you decide when you sign up*. When someone leaves you a message, e' alert you via email, pager or cell phone. You get free access to your messages from an world via the web at evoice.com. And you can leave a 1-minute personal greeting letti callers know you're not able to answer their call*.

(*=personal greeting and ability to choose number of rings available for eVoice Premiur subscribers only.)

Signing up for eVoice only takes a few minutes at evoice.com.

[Ba

What else do I get with eVoice?
Check your messages anywhere. With eVoice, you can check your messages by phone.

or have them sent to you as email. Try doing that with an answering machine!

Free long-distance messaging between eVoice subscribers. You can also send voice n other eVoice subscribers anywhere - free. Just call an eVoice access number, enter the number and leave a message. Wherever they are, they can call or check the web to lis message. Only eVoice has the features to keep you in touch with the world.

Forward messages from the web: You can forward a voicemail message to anyone wit address. From your eVoice inbox, click the forward button and we'll send the message RealAudio attachment along with your personal note.

eVoice All Access - combine your wireless phone with your home voicemail: As an e Premium subscriber, you can add a wireless phone to your mailbox for only $2.95 more Now instead of checking voicemail from two places, you can receive all your messages convenient place.

Message broadcasting: eVoice subscribers can send a message to up to 20 subscribers called QuickDial. It's great if you want to invite people to get-togethers or make an an Tell everyone with one call!

Phone message management keys: Using any touch-tone phone, you can save, skip, f rewind, or delete messages at any time. You can even forward them to other eVoice s reply directly to messages sent by other eVoice subscribers without paying long-distanc

[B

Where is eVoice available?
eVoice is available throughout most of the US and parts of British Columbia. You can c messages anywhere on the web or in your email and from practically anywhere using o numbers. When you start to sign up for eVoice service, you give us the phone number y answer. We'll tell you immediately if eVoice is available in your area. If not, check bac adding service in new areas all the time.

[B

Can eVoice answer my wireless phone?
Yes, eVoice customers in several sections of the United States are now able to add a w to their eVoice Premium mailbox, so that one mailbox answers both phones. It's called Access, and we're looking to roll this out to the rest of the US very soon!

Back to Top

"Voice-ASP, White Paper Technology & Processes," eVoice,
December 13, 2000

eVoice™

Voice-ASP

White Paper
Technology & Processes

1 Introduction

This White Paper discusses the underlying technology and processes of the eVoice service, focusing on the advantages of eVoice compared to other voicemail and messaging systems. eVoice's ASP-program helps Service Providers (Wireless, Long Distance, CLECs, ISPs, Voice Portals, etc.) deliver next generation enhanced communication services in a reliable and scalable fashion.

2 eVoice Voicemail

eVoice is a nationwide voicemail service that answers home, small office and wireless phones. Calls to the answered phone numbers are forwarded (on Busy or No Answer) to eVoice. If the caller leaves a message, the subscriber is then notified via e-mail and wireless notification and can access the voicemail via phone, the web or e-mail.

3 Service Setup

One of the main ideas behind eVoice's service is to simplify the previously complex call forwarding service setup and provisioning process for the subscriber as much as possible. eVoice streamlines the process by handling all required contact with the subscriber's Local Phone provider. This increases both the sign-up and retention rates.

3.1 RBOC Relationship

eVoice has spent several years developing close relationships with each of the Regional Bell Operating Companies (Bell Atlantic, Bell South, Ameritech, Southwestern Bell, US West and Pacific Bell) and GTE (the term "RBOCs" will be used for this group). The development and maintenance of these relationships allows eVoice to directly accept and process orders from subscribers, eliminating the need for the subscriber to contact the RBOCs directly. eVoice is constantly enhancing its provisioning capabilities (service setup), and additional Carriers (landline and wireless) will continually be added.

3.2 Billing

eVoice's relationships with the RBOCs also include billing arrangements. In most cases, eVoice handles both the set-up and monthly fees that RBOCs typically charge for Call Forwarding. This prevents the addition of fees to the subscriber's RBOC bill, which limits confusion for the subscriber, and reduces customer service calls to the RBOC.

3.3 Order processing

eVoice receives customer orders through both an automated telephone interface and web-based registration forms. These orders are directly forwarded to the correct RBOC using scalable electronic interfaces. eVoice has developed a customized automated interface for each RBOC, to enable timely and efficient processing of

eVoice's large daily order volumes, which currently exceed 2,000 orders per day per RBOC.

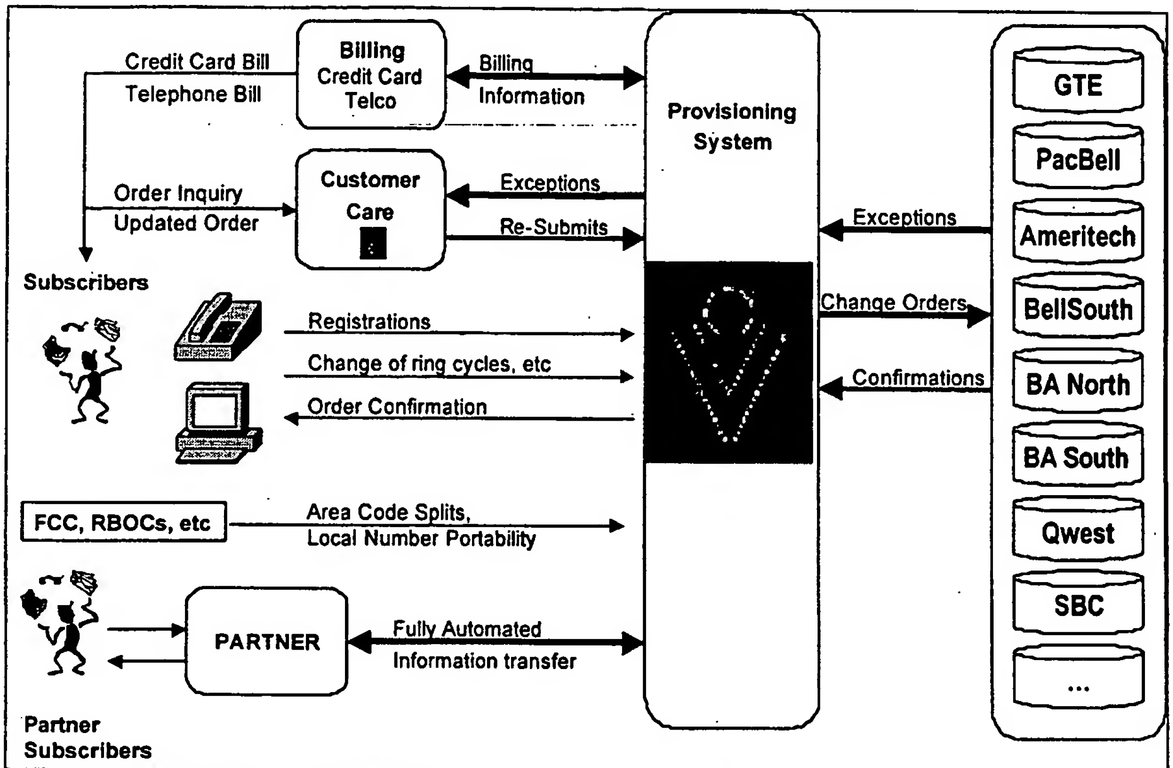


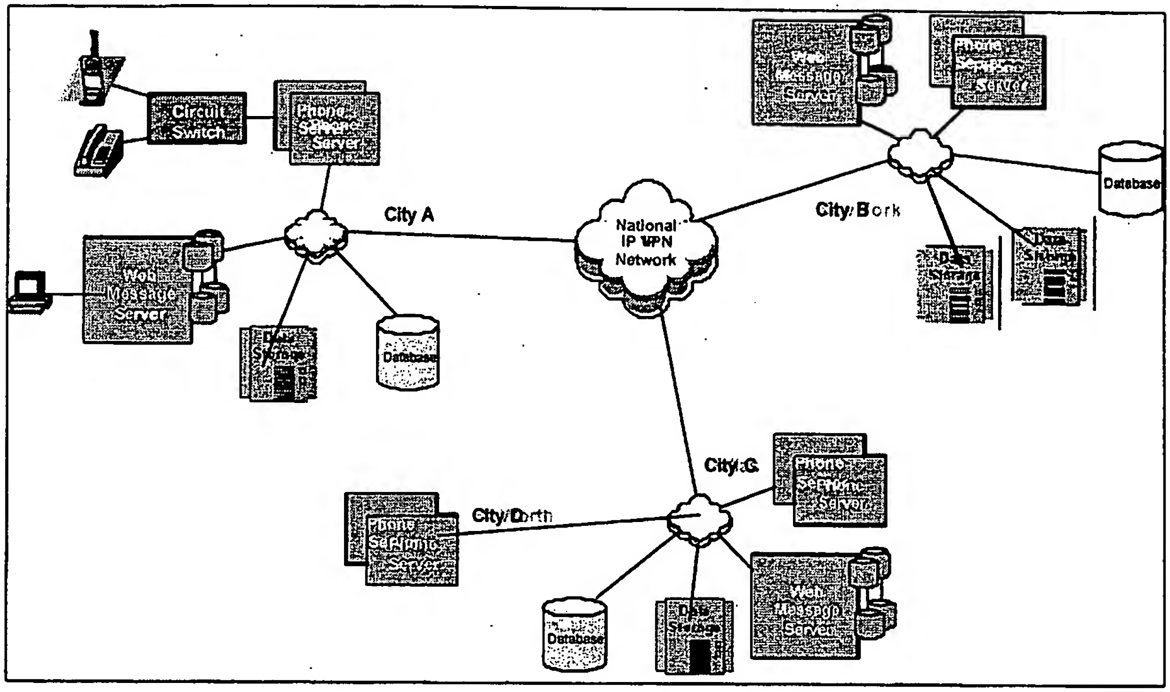
Figure 1 - eVoice Provisioning System

3.4 Exception handling

eVoice has a distinctive, knowledge-based method of handling returned orders, or "exceptions." An RBOC might reject an order or return it for further processing for one of several reasons, e.g.:

- The official number plans provided by the RBOCs are often not fully up-to-date (i.e. do not reflect numbers that have been resold), which may result in an order being sent to the wrong RBOC.
- The customer may have special features enabled on the line that conflict with the new service order.

eVoice has the unique capability to handle these exceptions by simultaneously contacting the customer to resolve the current issue and providing feedback to a rule-based database that is used for order acceptance. eVoice can thereby offer a complete provisioning solution, avoiding the need to train Customer Care staff in RBOC exception handling.



4 Telecom Network

eVoice is the pioneer in deploying IP-enabled voicemail. The path to building out the eVoice network has been much like that of the early ISPs, who had to build their own POPs. Similarly, eVoice has undertaken the process of building and operating an extensive Telecom network. The network has been designed to meet the highest standards of reliability and scalability, while also taking advantage of the low-cost aspects of IP-technology.

Figure 2 – Schematic System Layout

4.1 Local Numbers

eVoice has deployed an extensive Local Number Network, ensuring that every call is always forwarded to a local number. For example, in the San Francisco Bay Area alone, eVoice has over 60 different phone numbers, enabling local coverage of each rate-center (there are often multiple rate-centers within each area-code). All calls forwarded from the called phone number to the voicemail service will be billed by the RBOCs as new calls. Therefore, it is important to have local numbers available, to avoid adding extra costs to the customer's phone bill.

4.2 POP vs. DID

While it is important to provide local coverage for each rate center, this must be balanced with the effort to minimize deployment and equipment cost. eVoice achieves this balance by using POP-technology, enabling eVoice to have exactly one number per rate-center. eVoice uses a Patented software algorithm to identify the incoming call by gathering the signaling from the RBOC, and can thereby open

the correct mailbox automatically. This allows eVoice to minimize the number of forwarding telephone numbers required (each carrying a monthly fee).

Without this sophisticated proprietary technology, competitors must use separate telephone number for each customer (DIDs) in order to identify the correct mailbox. This is an expensive solution, since each number carries a monthly fee. In addition to being expensive, this solution is also not scalable because telephone numbers are a scarce resource, especially in urban areas. Numbers are therefore often obtained in low-density areas, forcing the calls to be forwarded outside the rate-center (and quite often outside the area-code), thereby adding toll-charges to the customer's telephone bill.

4.3 Transport Cost

The POP-network architecture allows eVoice to benefit from the advantages of IP-technology. Message are converted and stored as packages at the closest POP and the message is then transported via IP, if a different POP is used for accessing messages (most messages are left and checked at the same POP; eliminating all need for transport).

Example: A message that is left in San Francisco and is accessed in New York requires only a couple of seconds to transport over IP, whereas conventional (TDM) networks require a managed and monitored long-distance connection for the entire duration (minutes) of the message.

This transport method also eliminates the quality problems that normally plague VoIP connections, which can be expensive to correct and control.

5 System Architecture

5.1 Centralized logic

The eVoice system is built around a centralized and redundant Network Operations Center. This NOC houses all system logic, routing tables, customer profiles, etc. This allows for easy management of the nationwide system, flexible scalability and rapid roll-out of new features.

5.2 Decentralized Data and Message Storage

Messages are stored at the edge of the system, in the POP-servers. This architecture eliminates the potential "single point of failure" problem that is normally associated with a centralized architecture. Each eVoice message is stored redundantly, and the system even recognizes "roaming" customers (calling into the system from different locations) and automatically stores messages at the POP-server closest to the customer.

Customer data is stored in regional and Master databases for both high reliability and fast service. The flexible, decentralized architecture allows eVoice, at the

carrier's request, to store the subscriber data on-site at the carrier's facilities in a mirrored Master database

5.3 Transport

eVoice transports non-real-time messages (IP-packets) between POPs (when needed) instead of streaming voice to a centralized storage place. This transport method is much less costly and more reliable compared to hauling VoIP to a central storage place. VoIP is a good alternative for real-time voice communications, but it has poor economics for messaging and requires extensive maintenance to guarantee the voice quality.

5.4 Mailboxes vs. Phone Numbers

Most voicemail systems are either one-dimensional, where the telephone number is equal to the mailbox ID, or hierarchical, where one telephone number can have several mailboxes (extensions). The eVoice system design allows the best of both worlds by creating separate dimensions for telephone number and mailbox, allowing for a very flexible service setup.

Products that provide voicemail for multiple phones in a single mailbox are now entering the market. This feature will become more popular as an increasing number of people have at least two phones, such as a wireless and a landline. Current solutions often only focus, due to system limitations, on combining business and wireless phones, both phones that are of single user type. The home phone, on the other hand, often has multiple users, and therefore requires extensions, a requirement that currently most systems cannot manage. eVoice's architecture can easily handle this type of configuration, providing both multiple phones and extensions simultaneously, whereas most other voicemail systems only can handle one dimension.

5.5 Notification

eVoice system includes a highly flexible Notification architecture, allowing for notification of subscribers via multiple media. The Notification services are IP-enabled, with the main notification methods being e-mail, SMS and pagers. The subscribers can choose to include a copy of the voicemail in the e-mail, as a Real Audio attachment. The Notification system is already prepared for other media, as Instant Messaging, WAP and Stutter dial-tone, and these methods can swiftly be integrated with the carrier's services.

6 Conclusions

eVoice is well positioned to be the leader in enhanced voice messaging solutions. The unique nationwide home answering capabilities combined with the flexible, reliable and low-cost architecture offer several compelling capabilities for carriers (wireless, IXC, DSL, ISP, CLEC, etc.). The eVoice service can be fully branded and modified to fit the Carrier's current service offerings.

For further information, please contact:

Johan Samuelsson
Director, Voice-ASP
(650) 330 3758
johan.samuelsson@evoice.com

"Voice-ASP, White Paper: Market Opportunities for Enhanced Voicemail," eVoice, November 10, 2000

eVoice™

Voice-ASP

White Paper: Market Opportunities for Enhanced Voicemail

1 Introduction

The enhanced telecommunications services industry is growing rapidly. The convergence of the Internet and telecommunications industries coincides with growing consumer demand for enhanced telecommunications services. This includes services such as call management, call return, caller ID, call completion, call waiting, call forwarding, and voicemail. Voicemail is one of the most valuable enhanced services on the telecommunications landscape today. This is illustrated by recent trends in the residential telecom market where rising prices coincide with increasing penetration, and on the wireless side where voicemail is becoming a required feature for a carrier. This paper discusses the advantages and the value of adding eVoice voicemail (home, small business and wireless) to a carrier's (Wireless, IXC, ISP, CLEC, Voice Portal) current service offering.

2 eVoice Voicemail

eVoice is a nationwide voicemail service that answers home, small office and wireless phones. Calls to the answered phone numbers are forwarded (on Busy and No Answer) to eVoice. If the caller leaves a message, the Subscriber is then notified via e-mail and wireless notification, and can access its voicemail via phone, the web or e-mail. eVoice has developed an automated registration and provisioning process that creates a seamless signup experience for the customer, eliminating all of the hassle associated with contacting the local telephone company.

3 Market Size

The market for these services is growing rapidly. International Data Corporation (IDC) estimates that the consumer voice messaging service market alone will grow from $1.3 billion in 1999 to $2.3 billion in 2003, driven primarily by new subscribers. Current market penetration for voicemail in the U.S. is only 17% and is projected to grow to 30% by 2003, according to IDC. The growth in the number of wireless phone and device users is also driving increased demand for enhanced telecommunications services. Approximately 125 million users in the U.S. subscribed to wireless services in 1999, and this number is expected to increase significantly to 207 million users by 2004, according to IDC. Furthermore, it is believed that enhanced telecommunications services are the most profitable part of most regional phone companies' revenues which also makes this an attractive market segment.

3.1 Challenges for Consumers

As the number of wireless, Internet, and telephone users has increased dramatically in the past few years, the consumer demand for enhanced telecommunications services has outpaced the degree to which these services are streamlined and integrated. Many consumers with more than one phone number currently need to access multiple voice mailboxes to retrieve their messages, which is costly and time consuming. The emergence of wireless phones with visual interfaces such as Wireless Access Protocol (WAP) does not fully address consumer needs. Wireless users, especially while driving, desire more convenient access to communications.

In addition, many consumers accessing the Internet use dial-up connections and have only a single home phone number, meaning they cannot receive incoming calls while they are online. Incoming calls go unanswered or are forwarded to voicemail for many such consumers, and callers typically have no way of knowing if the person they are trying to call is online. This creates a significant consumer need for the type of call control that eVoice provides.

3.2 Challenges for Carriers

Recent deregulation of the telecommunications industry has greatly reduced barriers to entry for telecommunications service providers. The elimination of these barriers has increased competition for Regional Bell Operating Companies (RBOCs) and created opportunities for new entrants such as Competitive Local Exchange Carriers (CLECs) and long distance carriers, also known as Interexchange Carriers (IXCs), to enter regional markets. Deregulation has also enabled relative newcomers such as Internet Service Providers (ISPs) and enhanced telecommunications services providers, such as eVoice, to enter the market.

The rapid changes in the telecommunications industry have created significant challenges for many of these CLECs, IXCs, and ISPs. In response to increased consumer demand for enhanced telecommunications services, many carriers would like to add new features and functionality to their offerings to distinguish their brand, reduce churn and add incremental revenue streams. However, adding incremental enhanced services to a carrier's offerings is both time consuming and costly, requiring the carrier to build new infrastructure, deploy new operations support systems, and develop expertise in Internet and voice web technologies. Its believed that many carriers will find it economically more attractive to purchase enhanced services from an Application Service Provider (ASP) on a wholesale basis and re-brand them as their own, than to develop them internally.

4 The Value of Voicemail

This section discuss the different ways carriers will benefit from adding eVoice voicemail to their current offerings.

4.1 Monthly Fees

RBOCs (Bell Companies + GTE in this paper) are currently offering Home voicemail at a price between $6-10 per month[1]. eVoice Home voicemail offers several enhanced features (Web access, e-mail notification, etc.) that makes it a superior product to the current RBOC offering, and thereby justifies a similar Monthly Fee. Customers prefer monthly fees with unlimited (or close to unlimited) service rather than measured services that charge on a per-message basis. Flat fees are also preferable from the carrier's perspective, because they tend to foster increased usage which enhances the bond between the carrier and the customer.

---

[1] *It is important to add the fees for the separate but required Forwarding feature when looking at RBOC voicemail pricing. These fees are typically not included in the quoted price.*

Example	
Monthly voicemail fees (RBOC prices)	$7-10
Set-up fee (RBOC prices)	$10-15

4.2 Increased Usage

The value of increased usage of the main service is most clear for wireless carriers, but Long Distance Companies (IXCs) will also get a boost in usage and thereby revenue. An often quoted example of the benefit of adding voicemail is the "*Zero calls vs. Three calls*" scenario. An unanswered call to a phone without voicemail will result in *Zero* completed calls, whereas by adding voicemail, there will first be a call for leaving the message, a second call for checking the message and finally a third call when the called party calls back the caller. This results in *Three* completed calls, all of them chargeable (although message leaving is usually free of charge). This is the main reason that many new PCS providers and overseas carriers have added voicemail as a Free-of-charge feature to their wireless services.

Adding more minutes of use in today's world of "big bucket" wireless plans is often mistakenly considered to be negative or neutral, under the idea that "there are plenty of minutes left in the bucket and more minutes will not generate any new revenue." This is not true, since there will always be some subscribers that will be "pushed" into the next bigger bucket or pay for minutes over the bucket-limit. Averaging this revenue over all subscribers will show that each added minute will generate revenue. It is also important to driving as much usage as possible from the local phone to the wireless, causing landline migration. The wireless competition will continue to drive larger buckets and lower prices. It is therefore important to fill up these buckets in order to keep the ARPU[2] steady.

Example - Wireless	
10 extra "return" calls per month at 3 min = 30 extra minutes	
Marginal revenue per minute– 15 cents.	
Additional Monthly revenue due to voicemail =	$4.50

4.3 Lower Churn

Controlling churn (customer retention) is today's most important business issue for Telecom carriers. Increasing acquisition costs combined with tighter margins require the carriers to extend the relationship with each customer as long as possible. Lowering the monthly churn for wireless carriers with just 0.1% will increase the lifetime revenue by more than $90[3]. According to IDC, "*the best way to reduce churn is to build a relationship with customers. One way of doing this is to bundle services*." Adding voicemail is one of today's best bundles, proven by the fact that

---

[2] *Average Revenue Per User*

[3] *Current average wireless churn is 2.2%. Data from DLJ "Global Wireless Communications Industry" report, Summer, 2000.*

most new PCS providers choose to include voicemail as part of the PCS service bundle at no additional charge to the consumer.Communication with the customer is also an effective churn reduction tool. Most telecom carriers only have bill inserts, or, even worse, a line on a credit card statement as their only communication vehicle. eVoice web-enabled voicemail has proven to be a great communication vehicle, allowing the carriers to communicate with the customers daily in a one-to-one dialogue, via the web in-box.

Home voicemail is an excellent addition to the service bundling because it is not directly associated with the carrier's main product. Cancellations will thereby affect not one but two important services, making it much more difficult for the customer to leave the service.

Example - Wireless	
Lower churn from 2.2% to 2.0% will add to lifetime revenue	$180

4.4 Lower Cost

eVoice voicemail is based on state-of-the art software running on high-capacity IP-based servers. This configuration will dramatically reduce the cost per customer compared with traditional voicemail systems with proprietary hardware and software modules. The outsourced model will also lower management and maintenance cost for carriers, without reducing the carrier's control and reliability. (See also Section 5.2, Pay-per-Feature).

4.5 Differentiation

eVoice's nationwide home answering capability and highly customizable access and notification options provides carriers with an opportunity to differentiate their products from those of competitors. The basic service alone will give carriers an edge, and eVoice's flexible platform allows for extensive integration with carrier's existing products which allows for the ongoing development of new innovative services. Telecom services are rapidly becoming commodities (the ISP-industry is a recent example). Differentiation will allow carriers to avoid the price pressure and remain competitive (AOL being the outstanding example from the ISP-industry).

4.6 Increased Web-traffic

Web access is a great complement to the traditional phone interface for voicemail. The web inbox integrates neatly with the carriers current web-site[4]. This will give a dramatic boost in traffic to the carrier's web-site. Customers will visit the site to check for voicemail several times per week, allowing for promotion of the carrier's other web-services such as e-commerce, e-customer care or up-sell of other products. This will help the carrier to achieve the cost savings that the addition of web-services promised to deliver.

---

[4] *eVoice will handle the integration.*

4.7 Add-on Sells

eVoice voicemail offers several opportunities for selling additional revenue-generating services. Multiple Phones, Extensions and Internet Call Waiting (ICW) are several of the many features that can be added to eVoice service, generating added revenue per subscriber.

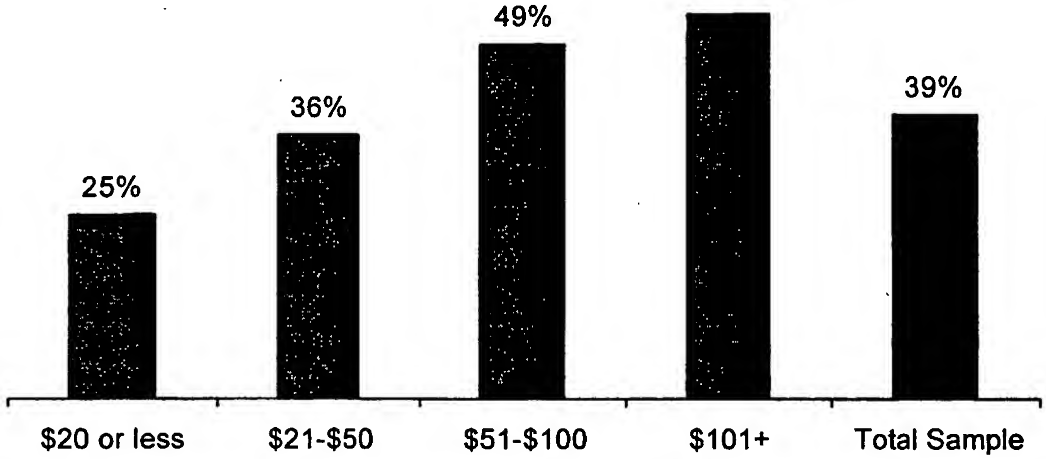
Example	
25% of subscribers with extension at $2.95/extension	$0.75
20% of subscribers with Multiple Phones at $4.50/extra phone	$0.90
20% of subscribers with ICW at $4.95/subscriber	$0.99
Average 20 minutes LD from voicemail platform at 10c	$2.00
Total extra revenue per subscriber	**$4.64**

4.7.1 Multiple Phones

Adding multiple phones into one single mailbox is a frequently required feature by today's busy consumers, that often are forced to check voicemails in multiple places. Several Bell Companies (Bell South, Pacific Bell and US West) have or shortly will launch this type of service. eVoice, with its home-answering capabilities, is the only company that can offer the same service for non-RBOC companies. 39% of wireless users indicate that they are interested in adding home voicemail to their wireless voicemail and 41% state that the availability of this service would influence their choice of wireless carrier[5]. This creates a sizeable market opportunity, and the interest increases among the higher-spending segments of the wireless customer base.

[5] *Market study performed by Insight Express, ordered by eVoice. Contact eVoice for more information.*

Figure 1 - Wireless Subscribers interested in adding Home voicemail to their Wireless service, segmented by monthly wireless spending (ARPU).



4.7.2 Extensions

RBOCs currently charge between $2-4 for each extension that is added onto a home voicemail box. The eVoice product that routes Multiple Phones into a single voice mailbox will require an enhanced usage of extensions for home phones, since home phones are often shared whereas wireless phones are private. 10% of eVoice current users are already using extensions for a single phone, a ratio that is increasing. The increased usage of extensions that comes with the Multiple Phones product makes the extensions feature that eVoice offers even more valuable.

4.7.3 Internet Call Waiting

Internet Call Waiting (ICW) allows customers to receive notification and to answer calls when they are on-line. ICW allows customers to avoid paying for a second phone line while still enjoying many of the benefits, reducing their overall telecom spending. This advantage is also available with the basic eVoice service, but ICW adds enhanced flexibility of how to handle incoming calls. Most ICW services are today in the range of $4-7 per month.

4.8 Platform for other services

The eVoice platform provides carriers with an excellent foundation for adding other telecommunication services . The frequent use of voicemail serves as the entry gateway for less frequently used but highly profitable services as Directory Assistance, Calling Cards, Voice Portals, etc. The flexibility and reliability of the eVoice platform allows for easy integration of other services.

4.9 *Example*

This example demonstrates the extra revenue that would be generated by adding home voicemail to a wireless carrier's existing product.

Each home voicemail customer will generate revenue in several categories: monthly fees, extra airtime and enhanced features (extensions and ICW).

Monthly Fee	$6.95
Extra Airtime	$4.50
50% penetration of extensions at $2.95	$1.47
20% penetration of ICW at $4.95	$0.99
Total added revenue per home voicemail subscriber	**$13.91**

Reducing churn from 2.2% to 2.0% will add an additional customer lifetime revenue of $180 (or $236 including the new Home voicemail revenue).

The take-rate for home voicemail is assumed to be 20% (using the rule of thumb that 50% of survey respondents that indicated interest will actually sign up for the service). Revenue will increase by $2.78 per subscriber, or by 5%[6] - a significantly increase in carrier's revenue.

5 Advantages with Voice-ASP

The Application Service Provider-model offers many advantages for carriers, and eVoice Voice-ASP brings these advantages for voice-services.

5.1 *Time-To-Market*

The initial integration of eVoice voicemail will be done in weeks instead of months of planning and installation of dedicated voicemail platforms. Add-on features can be added in days.

5.2 *Pay-per-Feature*

One of the greatest advantages with ASP is the possibility for the carrier to pay per subscriber and per feature. There is no need to build and pay in advance for capacity that will mostly be unused, or to upgrade the whole platform for enhanced features that only a small percentage of the customer base will use. EVoice offers a pay-as-you-go model where the carriers only pay for the services and feature they actually use.

[6] Based on an average ARPU of $55 for wireless carriers – DLJ's "The Global Wireless Communication Industry" Summer 2000 (Dick Tracey).

6 Value per Category

6.1 *Wireless*

Because the eVoice solution enables wireless providers to combine a user's land line voicemail with wireless voicemail, wireless users with eVoice service will use their wireless phone more often (to access their land line voicemail), resulting in increased minutes of use (MOU). As eVoice extends the capabilities of the platform in communications, content and commerce, this will further drive increases in MOU. These capabilities could provide a significant source of additional revenue and service differentiation and help reduce churn for wireless providers. In addition, the web features should attract users to the wireless carrier's website, providing opportunities for up-selling.

6.2 *Long Distance*

With the continuing deregulation of the telecommunications sector, RBOCs are beginning to provide long distance telephony service and are aggressively pursuing the $40 billion U.S. long distance market. Long distance carriers, also known as Interexchange Carriers (IXCs), resell local phone service, but do not have the right to resell the RBOCs' voicemail service. This inability puts the IXCs at a competitive disadvantage. By partnering with eVoice, IXCs will be able to provide customers with a more complete, competitive telephony solution. Furthermore, eVoice voicemail solution is superior to that of the RBOCs, and that this will enable IXCs to not only reduce churn, but also to increase their monthly billing revenues and their market penetration.

6.3 *CLECs*

Since deregulation in 1984, a large number of CLECs have been formed to compete for local service. CLECs have primarily been providing call completion services, and typically do not derive much revenue from enhanced services. In order to more effectively compete with the RBOCs, CLECs will need to offer more enhanced services, such as voicemail. By offering eVoice voicemail service, not only will CLECs have an additional revenue stream, but they will also have a superior service versus the RBOCs, thus improving customer acquisition and reducing churn.

6.4 *The ISP, Instant Messaging (IM) and Internet Portal Markets*

ISPs and online portals have strong customer bases with high usage, but limited presence in the voice market. Several of the online portals have added "voice chat" capabilities to their instant messaging products and also launched "PC to phone" capabilities. These portals do not typically have access to their consumers' home phones, and therefore, have not offered solutions that generate significant revenue from these home subscribers. eVoice enhanced messaging, IM and Internet Call Waiting capabilities can provide a compelling product for these ISPs and portals which should allow them to bypass the local phone companies. eVoice automated provisioning will allow ISPs to deliver advanced features such as Internet Call Waiting, so that users may manage inbound phone calls while staying connected on their dialup line. Also, web-based inbox provides web portals with an opportunity to

increase stickiness by enabling users to retrieve their voicemail messages and perform other communications functions from their website, every day.

6.5 Voice Portals

Advances in voice recognition technologies are enabling the emergence of "voice portals." A variety of new companies in the enhanced telecommunications services market are providing such useful content; however, companies that enter this market face two primary challenges: customer acquisition costs and telephony network costs. eVoice nationwide network and installed base of customers position address these issues. By integrating voice portals into the network, eVoice can significantly reduce the voice portal's telephony network costs and customer acquisition costs. eVoice can also provide low-cost user acquisition to voice portals by promoting the voice portal to the base of daily voicemail users, offering the convenience of direct-connect to the voice portal from the eVoice platform.

6.6 Summary

The table below summaries the opportunities for Value Creation for each segment.

	Wireless	IXC	CLEC	ISP	Voice Portal
Monthly Fees	★	★	★	☎	✉
Additional service usage	★	☎	✉	✉	✉
Lower Churn	★	★	★	★	★
Lower Cost	☎	☎	☎	☎	★
Differentiation	★	★	★	★	★
Increased web-traffic	☎	☎	☎	★	★
Enhanced features	★	★	★	★	☎
Value Creation:	*★ - Significant*		*☎ - Healthy*	*✉ - Some*	

7 On-line Marketing of Voicemail

Voicemail is a traditional telecom product that works well with normal marketing methods. However, the web-based inbox and registration have allowed eVoice to rely heavily on on-line when marketing its own brand of voicemail, with great result. It is therefore recommended to tap into on-line marketing as much as possible when designing marketing plans.

An even more important fact is that online marketing perfectly targets users of web-based voicemail. Once a customer has signed on for eVoice web-based voicemail,

the preferred marketing vehicle for that customer will be on-line, thereby significantly lowering future acquisition cost.

8 Conclusion

eVoice is uniquely positioned to offer carriers new profitable revenue streams, and at the same time lower churn and enhance differentiation. eVoice flexible platform allows for speedy and simple integration with the carriers current services. voicemail is already today one of the most valuable voice-services, and with eVoice, its full capacity is un-leashed.

For further information, please contact:

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“Audio Digitizing Process,”
TalkBank,http://www.talkbank.org/da/audiodig.html

TalkBank Audio Digitizing Process

What you will need:

1. Audio source (DAT, casette, minidisk, or reel-to-reel)
2. Power supply for the audio source
3. Mini-mini or RCA-mini audio cable
4. Headphones or speakers
5. Mac or Windows computer
6. Digitizing software:

- For older Macs: Sound Edit 16 (http://www.macromedia.com/software/sound/)
- For Windows: CoolEdit (http://syntrillium.com/). Also, Windows provides sound recording in the System Accessories panel for free, although this feature has some limitations.
- For OS X: Peak 3.0 (http://bias-inc.com). There are also several freeware and shareware recorders, but they have various limitations.

Connecting the output source to the computer:

The first step in digitizing a sound file is to connect the audio source (DAT, casette, minidisk, or reel-to-reel) to the computer. We will refer to the input audio source as a tape recorder. However, DAT, casette, minidisk, and reel-to-reel all work in essentially the same way. Then connection is usually done by connecting a mini audio cable from the headphones output on the tape recorder to the sound input on the computer or sound card. For better sound quality we used to recommend going through the line output and not the headphones. However, newer sound cards make it possible to use the headphone output. If you are able to get a good level of audio input from the headphones jack, you can skip reading the next section.

Using a mixer:

Some combinations of older hardware (often reel-to-reel and sometimes DAT) and older sound cards cannot make good use of the output from the headphones jack. In these cases, you need to use the line output from the audio source. The problem with using the

line output is that you often cannot control output level or volume (although some audio sources have a control for this). If the output level of the line output of your machine does not match that required by the computer, you will overdrive the input and get bad results. Newer sound cards seem to have solved this problem and it is usually possible to use line input for recording.

If this situation arises, you will need to use a device to control line output level. Another reason to have a device to control line output level is that, if the tapes are of poor quality or contain a large range of volume levels you will need to adjust the sound manually at various points in the tape in order to prevent clipping. The best solution is to use a mixer. The Studio Master 42DC which costs $65 can be purchased from Full Compass Audio (800 356-5844). Audio technicians tell us that mixers are better than amplifiers. However, we have not noticed an audible difference. Running the tape recorder through an amplifier allows volume control while maintaining the sound quality.

Setting up your connections in the simplest case

1. Get a mini-mini stereo cable.
2. Plug one end into the headphones output of the taperecorder.
3. Plug the other end into the computer's microphone input jack
4. Plug headphones or speakers for monitoring into the computer's headphones jack.
5. Open up SoundEdit16 of CoolEdit – you don't have to start recording yet. Just have it open.
6. Start playing the tape and listen to the sound. Observe the sound level and make sure that it does not go into the red zone. This feedback is in the Levels window in SoundEdit. It is less clear in CoolEdit.
7. You can control the sound level with the tape recorder volume controls.

Setting up your connections when you have an amplifer or a mixer

1. If you have a mini output on your taperecorder, you need a mini-dual-RCA cable
2. If you have a dual RCA output on your taperecorder, you need a double RCA cable.
3. Plug the correct end of one of these cables into your tapecorder's line output.
4. Plugs the other end of one of these cables into your mixer or amplifier's line in.
5. Now you need another cable to connection to your computer. This should be a dual RCA to mini cable.
6. Plug the one pair of RCA plugs into the mixer's line output.
7. Plug the mini jack into the computer.
8. Plug your headphones into the computer's headphones jack.

9. Open up Peak, SoundEdit16 or CoolEdit-- you don't have to start recording yet. Just have it open.
10. Start playing the tape and listen to the sound.
11. You can control the sound level with the amplifier or mixer volume controls.

Sound Settings

Sounds settings for your card change with each version of the operating system, so we cannot give instructions that will be valid for all users. Basically, you just have to read your computer manual for this. Make sure that you have the input and output settings adjusted to tape input from the external audio source and to play back to speakers or headphones if you want to monitor the recording.

Using Sound Edit 16 and CoolEdit

We describe the recording process here for SoundEdit 16. CoolEdit and Peak work in much the same way. For CoolEdit, try to ignore the complex interface and just focus on the recording buttons down at the bottom left of the screen. For SoundEdit 16, recording will default to 44,100 Hz, 16 bits, mono, with no compression. This is OK for recording. However, you will probably want to save the file at 22,050.

The Controls window

The Controls window in Sound Edit 16 is used to record sound, play a sound file and stop or pause the recording. You can also use the controls pull down menu at the top of the screen for the same options.

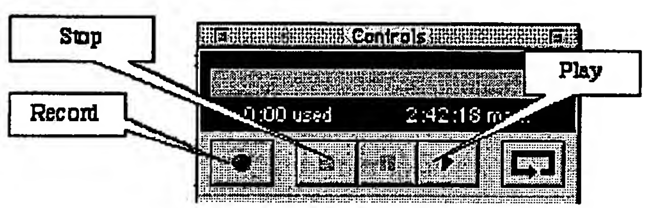


Figure 2: Control Window in Sound edit 16

Monitoring Sound levels

The Levels window allows you to monitor and control the recording and playback levels in Sound Edit 16. For the best results, in the recording window, make sure to set the L/R volume to –16. This setting is recommended by Sound Edit for obtaining the best over all sound quality.

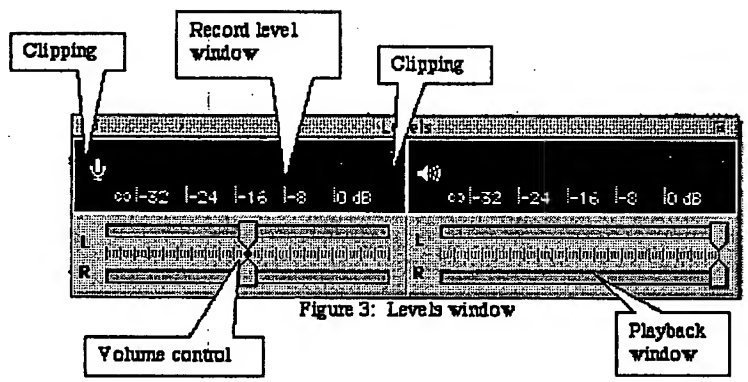


Figure 3: Levels window

If a red dot or red block appears in the recording levels area the sound is being clipped. Clipping is defined by Sound Edit as the amplitude of a sample exceeding the quantization range. For example, if a child is sitting to close to the microphone and starts to scream the top and bottom of the wave form will be cut off. This results in a very poor digitized sample.

In order to control for clipping, adjust the volume of the source (i.e., amplifier, mixer, or tapecorder) until the red light or dot no longer appears in the Recording level window. You can also control the volume using the input or volume controls in the Recording level window. Sound Edit does not recommend this method because the sound quality may be sacrificed.

Creating a new sound file

1. Load your cassette into the tape recorder.
2. Plug your headphones or speakers into the headphone jack in the computer.
3. Adjust the volume on your tape recorder.
4. To check volume start the tape without recording in your computer's program.

5. Adjust volume as needed on both sides.
6. Remember to rewind the tape to the beginning before going to the next step.
7. Press the play button on the tape recorder.
8. Press the record button in the Sound Edit control window.
9. Make sure to monitor the sound in the record Levels window for clipping.

"Supplemental Report to Diary 53, Networking the Sound Digitizing Device," Old Colorado City Communications and the National Science Foundation Wireless Field Tests, October 20, 2002, Lansing, Michigan,
ttp://wireless.oldcolo.com/biology/ProgressReports2002/Progress%20Reports2002/53SupplementalReport(10-20-02).htm

Old Colorado City Communications
and the
National Science Foundation
Wireless Field Tests

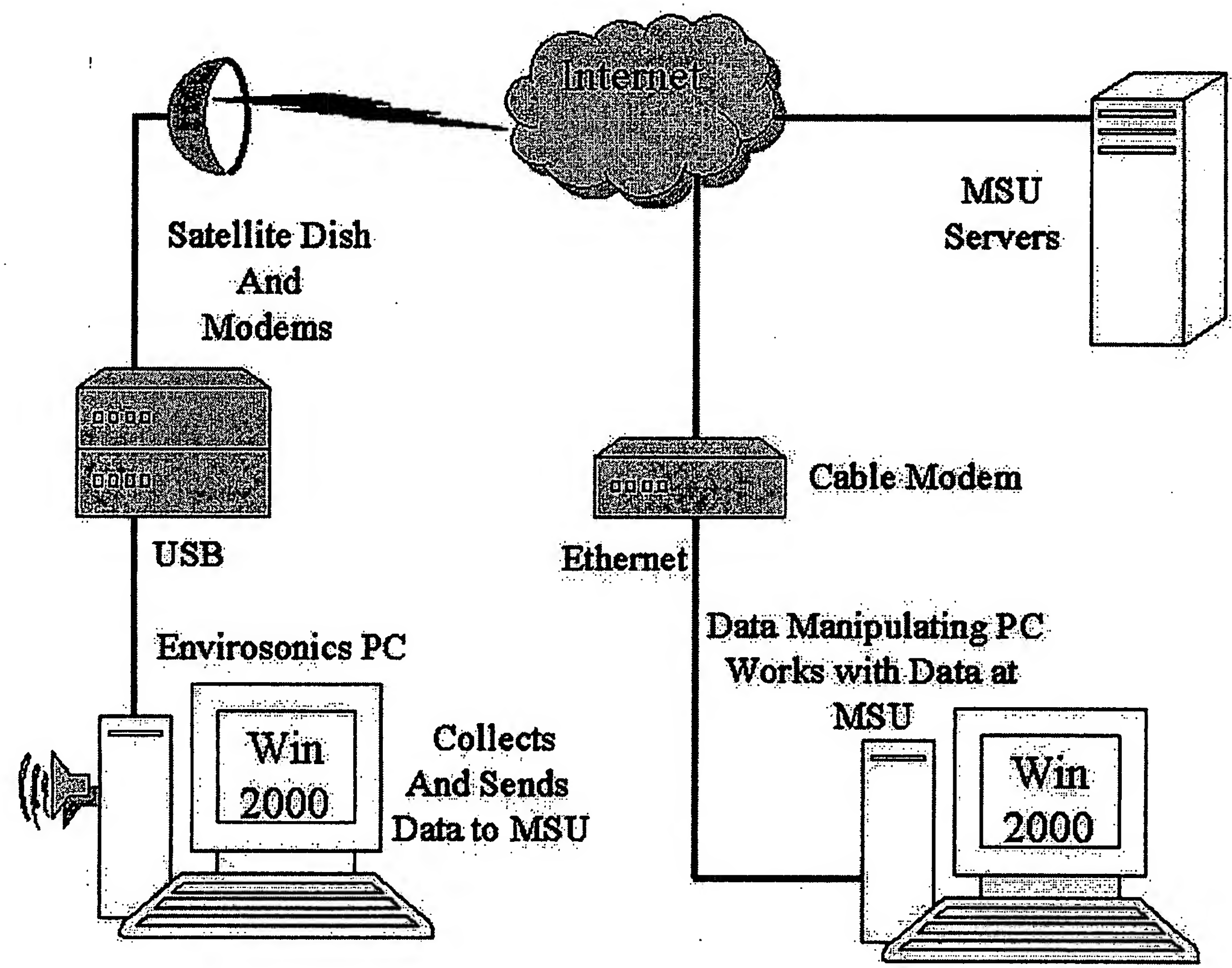
Supplemental Report to Diary 53

Networking the Sound Digitizing Device

October 20, 2002

Lansing, Michigan

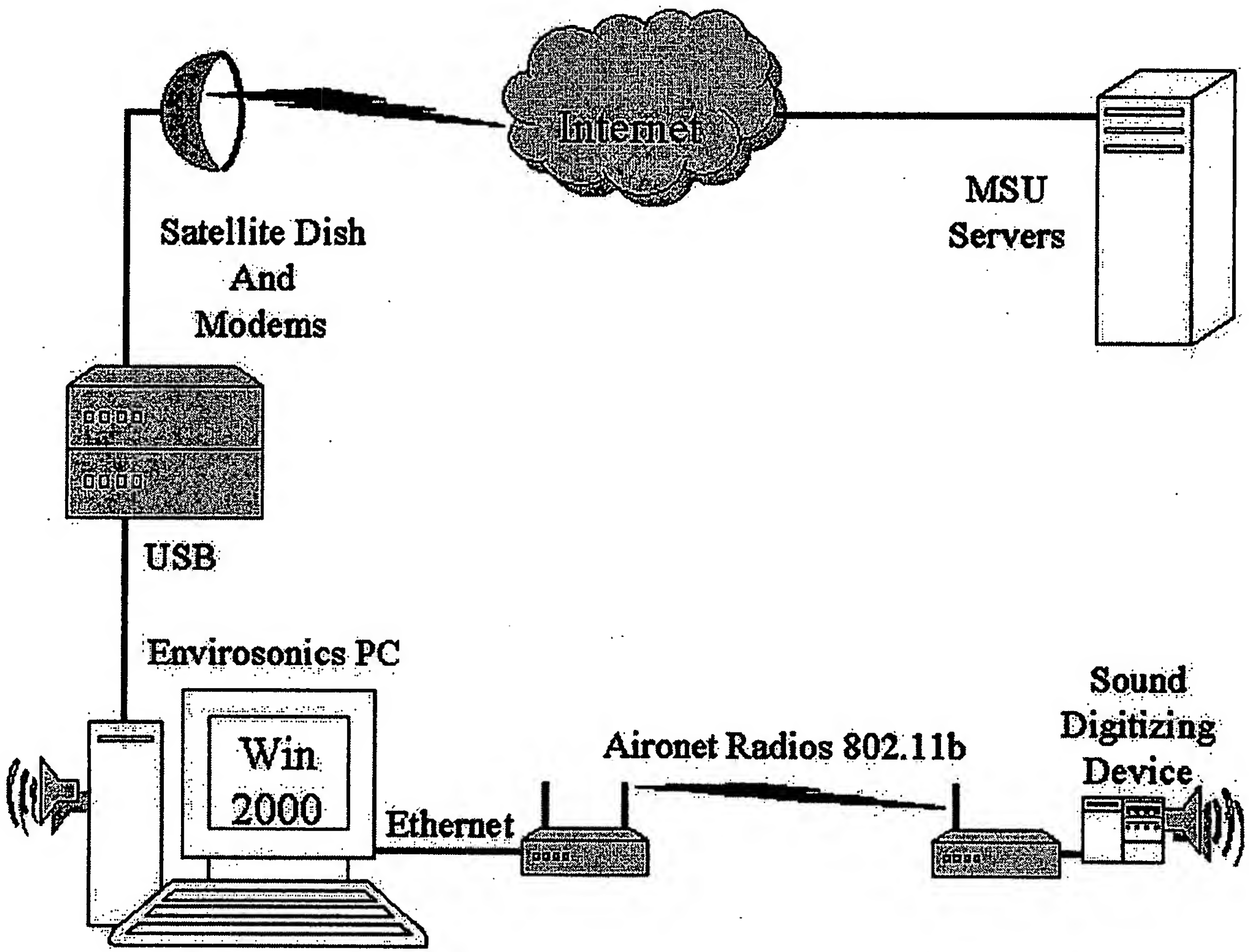
The networking portion of the prototype Sound Digitizing Device installation is explained in the following text. Our goal included working within the operational confines of the existing network architecture, currently supporting previous envirosonics related research. The existing network included a Satellite feed via a DirecPC two-way communications Internet link and a two-way communications Internet link via a local cable modem service. See the network diagram below:



The desired method of communications to the MSU server was through the DirecPC satellite system to best emulate a field installation. However, the satellite system was not designed to provide the generic network connectivity we needed. The satellite system we were working with was designed to work with a standard desktop PC running proprietary satellite networking support software through a USB interface. The satellite system supported the sound capturing PC being used as office level model for this research, and could not be disturbed. The network IP from the satellite feed is also a non-routable private IP address, so getting to the Sound Digitizing Device system for updates and changes would not be easy. Networking the Sound Digitizing

Device prototype on the satellite feed would be a difficult and messy networking job, further complicated by the satellite feed line to the PC being based on a USB link rather than Ethernet.

To provide the digital audio data to the satellite, we would have to dual home the win 2000 PC and utilize an Ethernet network card to provide a network link between the win 2000 system and the Sound Digitizing Device. See Below:



This was the network topology after our first visit to Lansing to install the SDD. It provided the connectivity between the SDD and the Envirosonics PC. However, it did not provide access from the Internet to the SDD directly.

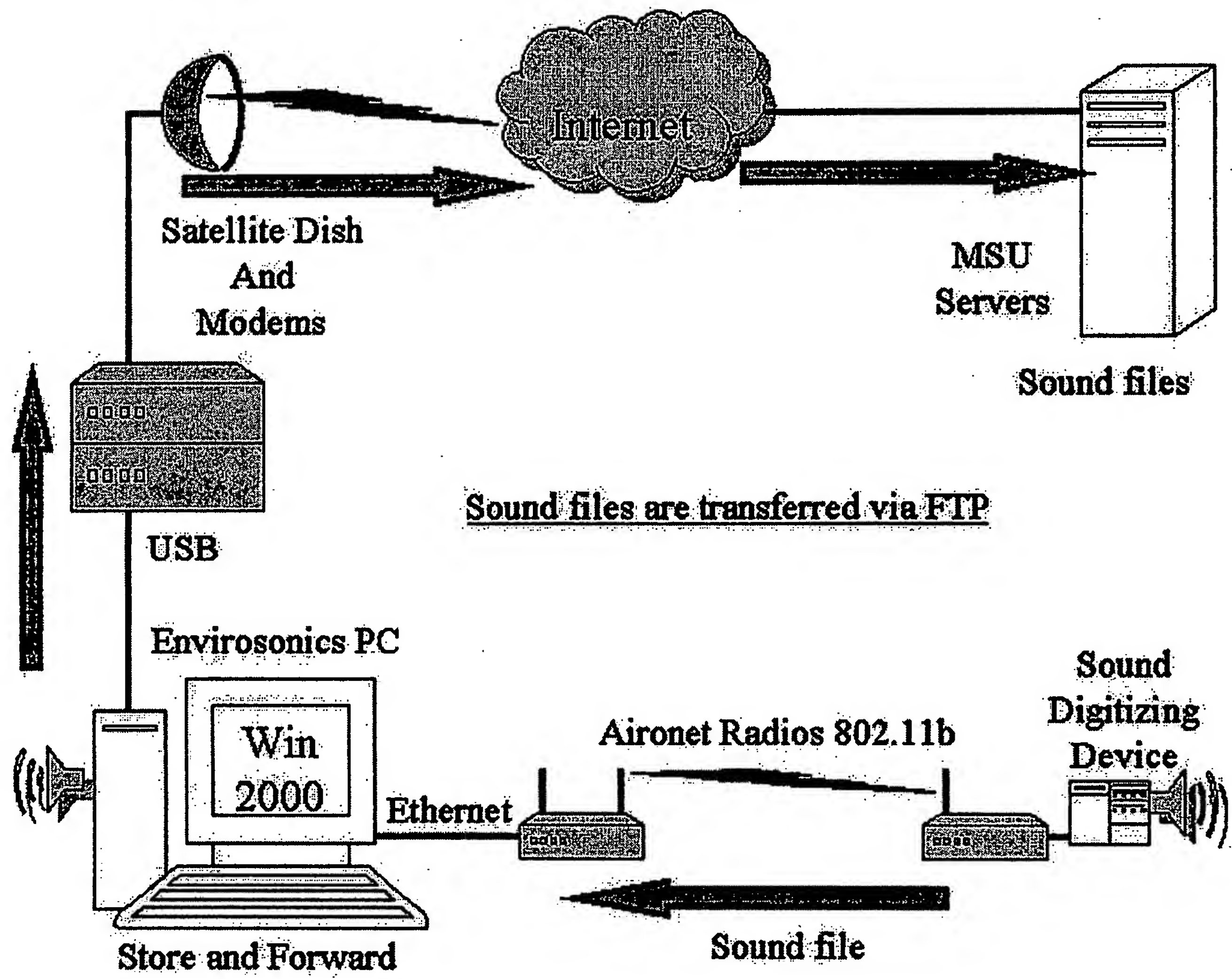
The flow of sound data starts at the SDD at the lower right. The sound is recorded digitally and stored locally on the SDD utilizing a program called LAME to rip files from the live-ice generated stream. The sound data is also provided as a live audio stream that can be monitored on the Envirosonics PC using a program such as WinAmp. However, the recorded file stored locally on the

SDD is transferred to the Envirosonics PC via the network link provided by two Aironet 2.4 GHz ISM band DSSS radios. Both the Envirosonics PC and the SDD connect to the Aironet radios via Ethernet. The file is transferred and named according to a scheme, which denotes the day and hour, but is completely configurable. A copy of the ftp'd file is then transferred from the Envirosonics PC to the Internet through the DirecPC satellite to an MSU server via FTP. A unique program, existing in the Unix environment for many, many years called "Expect" is utilized to mimic a user typing command line arguments into a program interface. In other words, "Expect" is a programmed virtual user inside the SDD that interacts with the ftp program to initiate the ftp session to the Win 2000 machine. An expect script is quite simple;

```
Send "FTP 192.168.0.1"
Expect "login"
Send "MyUserName"
Expect "password"
Send "MyPassword"
Expect "ftp >"
Send "bin"
Expect "binary"
Send "put stream10-21-1230"
Expect...
```

The expect program is able to interact with a great many programs, making it a very powerful tool to perform house-keeping within a computer system, special tasks, such as we are using, and can even perform decision making in many forms.

The network path of the sound files transferred via ftp is shown below.



A few problems remain after the implementation shown above:

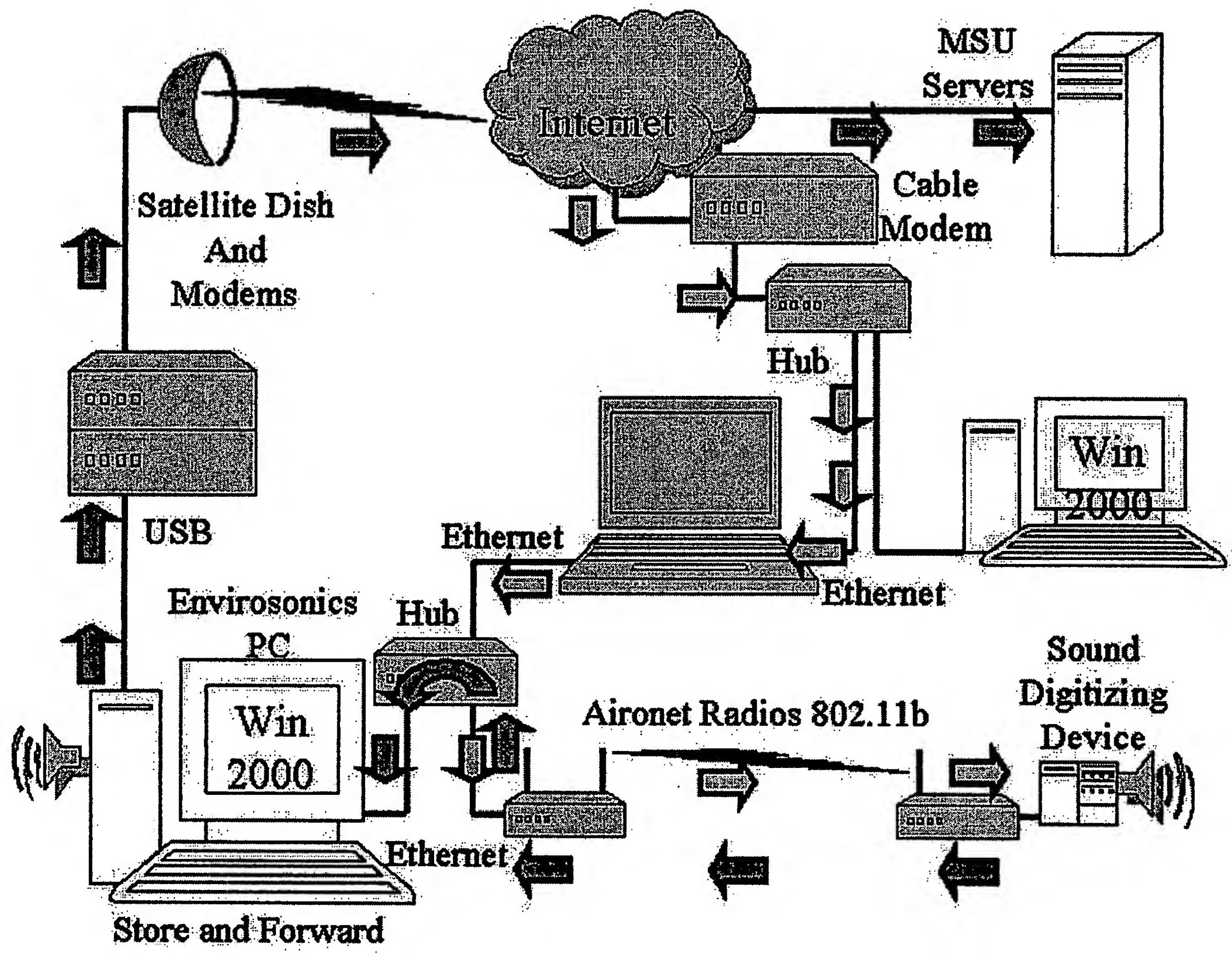
- **No remote access could be provided through the DirecPC Satellite connection for updates or adjustments.**
- **No streaming outside of the Envirosonics PC could be accessed.**
- **Win 2000 machines do not make the best Internet gateways, especially when one interface is a USB port talking to a satellite via proprietary means.**

· The IP addresses used within the DirecPC network are not routable from outside the DirecPC network. This could be alleviated via a virtual tunnel to a trusted remote machine.

To alleviate the problems as easily as possible, we looked at the Cable Modem side of the network. Our solution to the problem was to install a gateway system that does NOT route packets, but rather has access to both networks. The gateway system chosen was an IBM Thinkpad system running Redhat Linux 7.3, using two Ethernet interfaces. One interface would connect to the Satellite network, and the other interface would connect to the Cable Modem network.

If you follow the yellow arrows, you will see the path used for remote management of the SDD, which allows connections from one network to the other network, but not in a routed fashion. You must log into the laptop acting as a gateway between the two networks to access the other network. If you follow the maroon arrows, you will see the ftp path for the audio files discussed earlier.

Yellow Arrows: Remote Management
Maroon Arrows: Audio File Transfer Path



To provide an added security measure, both the laptop in the middle of the picture above and the Sound Digitizing devices utilize Secure Shells (SSH). The secure shell is accessed via a program called "Putty" which exists in the public domain and runs under Windows. You open up the SSH program "Putty" and select the SSH connection to the appropriate IP address. After authentication, the SSH program allows access to the system as if it were a telnet session. Most other sockets or ports on the Thinkpad laptop and the SDD have been shut down to provide fewer opportunities to be hacked.

The laptop was picked for the gateway for several reasons, such as:

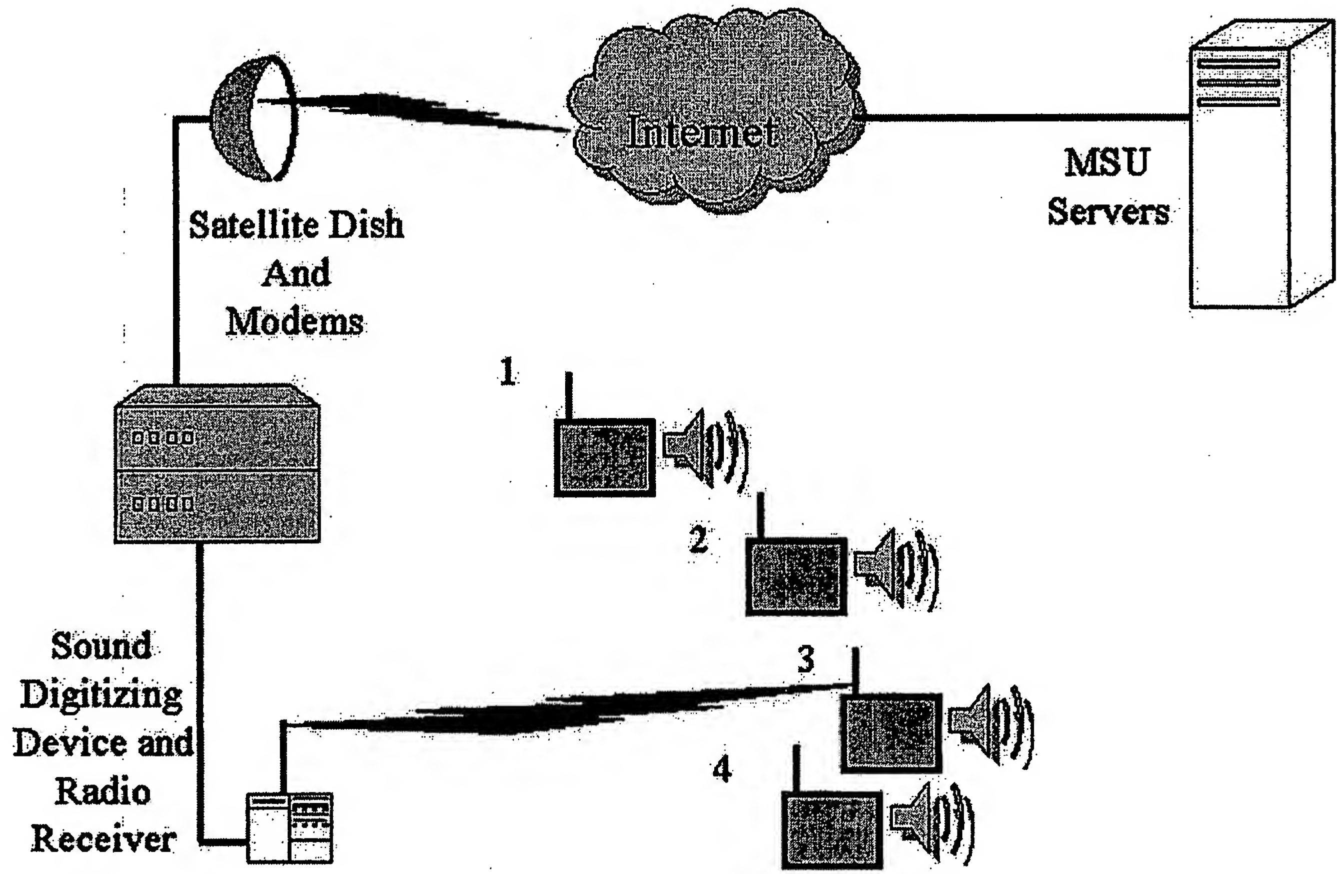
1) Linux is known to run well and install easily in the IBM Thinkpad systems, and since this was a remote install with a time crunch, things needed to go smoothly.
2) A laptop has automatic battery backup when the 110V AC power fails.
3) The size is small and the CPU power more than adequate for the intended use, as well as possible future streaming use.
4) Linux is very network friendly, easily configured multi-network operating system, so this provides a reliable access method in a true multi-user environment.

The SDD system provides several functions over and above the command line interface. Since the typical biologist is not a network guru, we have provided a "point & click" interface via a web-based tool. This tool is called "Webmin" which is easily configured to provide a point & click interface to many lower level Linux programs. From this interface you can manipulate IP addresses, or which programs run or listen to sockets, configuration of programs, configuration of users, etc. It makes Unix administration as easy as that of a Windows based machine. Webmin has the added ability of being changed when something does not do what you would expect, or you could write your own simple tool to provide the manipulation of some parameter not readily offered in the pre-packaged Webmin interface.

Another web-based interface is provided by the streaming software. By accessing a specific port on the SDD via a standard browser, you can look at the history of recorded digital sound tracks previously transferred to the server system. This interface can provide another way to access the data recorded by the SDD for use other than the routine use it was designed for. You can also force a recording if an event was happening in the area that would provide interesting data, such as the sound of a storm, a migration of birds, or man made sound events.

Keep in mind the complexity of the network shown in the previous network diagrams does not reflect a field deployment of a SDD, but rather to fit the SDD prototype within the confines of an existing network. The actual network implementation should be designed in advance to facilitate the needs of the device and all humans that will support the system, while targeting simplicity for a main goal. Below are two examples of what a field deployment might actually look like:

Example 1: Using "Smart Sensors" with IP all the way to the sensor:



A cost effective approach using "less smart sensors" which send audio
Via FM or some means preserving fidelity to the Sound Digitizing Device
In a time multiplexed method

In the design immediately above with "Less smart - smart sensors", another not-so-obvious parameter was alleviated: *Power*. By using very simple circuits as the acoustical sensors, and the digitizing done in only one place, the need for fairly large amounts of power is greatly alleviated. To follow this power reduction effort even further, a far more simple SDD could be designed such that the digitization takes place within a DSP (Digital Signal Processor) Integrated Circuit (IC) programmed to perform this function and requiring far less power than any off-the-shelf device we have available to us at this time.

Conclusion:

The network additions performed to facilitate this prototype installation were quite creative and reaching for opportunities to manipulate the network as passively as possible, while providing the connectivity desired. The operational design from the SDD viewpoint reflected a desire to provide capabilities we would like to have in a true field installation designed for envirosonics research. We achieved all of the goals for the prototype, some of which were formal requirements, and others which were "would it be neat if...." desired modes of operation. We have shown several network designs, with examples of what would be far more simplistic in implementation, along with ideas of the next generation Sound Digitizing Devices that would facilitate envirosonic research completely.

Michael Willett
Senior Technical Assistant and Collaborator

PREVIOUS

Chronological Progress Diaries | Purpose | Scope | Participants
Project Plan | Regulatory Issues | LTER | Top Page

“Macromedia SoundEdit 16 Support Center-Working with Other Programs, What is Shockwave Audio Streaming?”http://www.macromedia.com/support/soundedit/how/shock/whatis.html

Macromedia SoundEdit 16 Support Center
Working with Other Programs

What is Shockwave Audio streaming?

What we mean by "streaming audio" is that the sound is delivered to you as it is being received from the web site that you are visiting. This is very different from downloading a file to your hard drive and then playing it after the entire file has been downloaded. The advantage of streaming is that there is no waiting (or very little) from the time you click the mouse until you hear the sound.

Audio is very information intensive -- it takes a lot of data to represent a sound accurately. On the other hand, even a very fast modem processes data at a relatively slow rate. (Think of it as a big stream of water being pushed through a tiny pipe.) Because of this, it is necessary to compress the audio so that it can be squeezed through the modem to be played back at an acceptable quality. Shockwave Audio uses very sophisticated mathematical analysis to compress the sound so that it can be represented by relatively few bytes of data. This much smaller data stream is sent through your modem, uncompressed in your computer, converted back into audio, and then played through your speakers.

Shockwave Audio is scaleable, which means that you can select the quality level to use for the audio. Be careful when making this choice -- a very high quality setting may contain too much information to squeeze through a modem in real-time. In this case, users may hear gaps in the playback. For example, if you want people with 28.8k modems to stream successfully, limit your SWA files to lower quality settings. However, if you know that the audio is intended for user's on a high-speed network such as a corporate intranet, you can use the highest quality settings with very good results.

"Chapter 3: Overview," last updated December 2, 1999,http://service.real.com/help/library/guides/g270/htmfiles/overview.htm

Chapter 3: Overview

Welcome to RealServer, the streaming media solution! RealServer streams audio, video, image, animation, text, and other data types. RealServer also allows you to grow with your changing needs. This chapter introduces RealServer concepts and features.

To begin serving right away, consult Chapter 1, "Quick Start".

What Is RealServer?

RealServer is software that streams media-both pre-recorded and live events-over a network. The client receives the media in real time, and does not have to wait for the clip to download.

Components of RealServer

RealServer software consists of the following components:

- **Executable**-RealServer's main software, called `rmserver.exe` for Windows platforms, and `rmserver` for UNIX platforms.
- **Plug-ins**-these files provide the functionality of RealServer's individual features. Because of this open architecture, third parties can create custom features, allowing you to extend the abilities of your RealServer.
- **Configuration file**-a text file, based on XML format, that stores all of your RealServer's customized information. The configuration file name is `rmserver.cfg`.
- **License file**-one or more files which control the features enabled in your **RealServer**.
- **RealSystem Administrator**-a Web-based console for customizing and monitoring your RealServer.
- **Tools**-additional software tools such as the Java Monitor, which allows you to view how many clips are being served at a given time, and G2SLTA, which broadcasts pre-recorded clips as if they were live events.
- **Other files**-depending on the particular RealServer package you purchased, your installation may have other files that perform additional functions, such as commerce or ISP hosting.

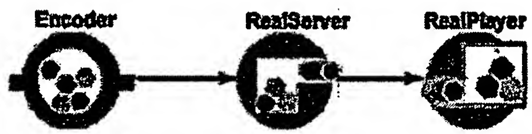
What is RealSystem?

RealServer is a member of the RealSystem G2 family of software tools. Three components make up RealSystem G2:

- **Production tools**-such as RealProducer Pro or RealProducer Plus that create media (such as audio, video, or animation)
- **RealServer**-which streams media
- **Client software**-such as RealPlayer, which plays the streamed media

The following diagram provides an overview of how RealSystem components work together.

RealSystem Components



Production Tools

The person who designs the content that you serve from your RealServer uses production tools to create the content. These tools convert audio, video, or animation to a data type format that RealServer can stream.

The content creator may additionally create a SMIL file to synchronize several clips in a single presentation. A SMIL file coordinates the playing and layout of media clips in parallel or sequence.

Since RealServer is able to deliver many formats, there are many tools that can be used in creating content. Production tools can optimize the material for delivery over the Internet, based on the content of the material and the expected capabilities of the users' equipment.

The content creator can prepare media clips in advance, or can encode a live event as it happens. In this manual, we use the generic term "encoder" to describe the software (such as RealProducer) that converts media or events into a format that RealServer can deliver.

RealServer

Just as a Web server delivers pages to Web browsers over the Internet, RealServer serves media clips, created with the production tools described earlier, to clients. It allows users to stream, rather than download, the media clips. By streaming the content, the user can begin to watch the clip almost immediately and does not have to wait for the entire file to download.

Client Software

A client such as RealPlayer plays the streamed media.

Other Software

In addition to the RealSystem G2 software, you may work with additional optional software, such as:

- Web server
- Web browser
- Firewalls
- Networking software
- Database software, if commerce authentication features are in use
- Ad server or services, if advertising features are in use

How RealServer Works

RealServer streams media to clients over networks and the Internet. It is usually employed in conjunction with a Web server. Some RealServer features can interact with third-party products to create specialized functions, such as report analysis.

Channels and Protocols

RealServer uses two connections, known as "channels," to communicate with clients: one for communication with the client, and one for actual data. The communication channel is known as the "control channel," since it is over this line that RealServer requests and receives passwords, and the client sends instructions such as fast-forward, pause, and stop. Media is actually streamed over a separate "data channel".

Every link to content begins with a protocol identifier, such as `rtsp`, `pnm`, or `http`.

RealServer uses two main protocols to communicate with clients: RTSP (Real Time Streaming Protocol) and PNA (Progressive Networks Audio).

Occasionally, RealServer will use HTTP for metafiles that point to RealServer content, and for the HTML pages served by RealServer (such as the Web-based RealSystem Administrator). It may also be used in delivering clips to clients that are located behind firewalls.

Within these channels, RealServer uses two other protocols for sending instructions and data:

- TCP-sends commands from the client such as "start" and "pause," and from RealServer to clients for information such as the clips' titles
- UDP-sends the actual data

See Chapter 9, "Firewalls and RealServer" for more detailed information on RealServer's use of ports.

Occasional Exceptions

Because many firewalls are configured to allow only TCP connections or HTTP traffic, you may need to make some adjustments to receive data from an encoder or to work with clients if there is a firewall between it and your RealServer. See Chapter 9, "Firewalls and RealServer".

Communication Between Encoder and RealServer

When the encoder connects to RealServer and sends encoded media data, it uses a one-way (UDP) connection to communicate with RealServer.

UDP Connection Between Encoder and RealServer

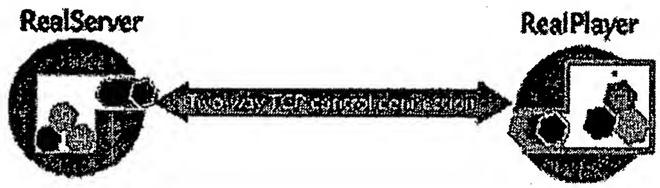
Some firewalls do not permit UDP packets, so RealNetworks encoding software such as RealProducer has a setting that uses TCP connections to send the same encoded media, since many firewalls allow TCP traffic.

TCP Connection Between Encoder and RealServer

Communication Between RealServer and RealPlayer

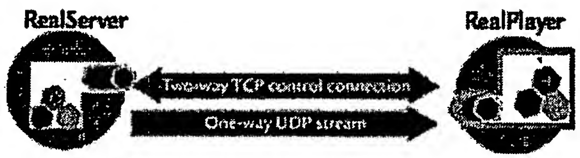
When the user clicks a link that points to a media presentation, RealPlayer opens a two-way connection with RealServer. This connection uses TCP to send information back and forth between RealPlayer and RealServer.

Initial TCP Control Connection



Once RealServer approves the request, it sends the requested clip along a one-way UDP channel.

UDP Data Connection



As it receives the streamed clip, RealPlayer plays it at high fidelity.

Streaming Delivery Methods

There are two main ways for controlling how a user experiences a clip:

- **On-demand**-like renting a video at a 24-hour video store, the clip is available to the user whenever she wants. The user can fast-forward, rewind, pause, and RealServer sends the right part of the clip. This type of clip is pre-recorded or pre-assembled.

- **Live**-like a live telecast of the Olympic Games, users tune in to the action that is happening now. A user can't fast-forward or rewind through the clip, because the event is happening in real time. To deliver content as a live event requires that there actually is a live event, and that you or the content creator have the software and hardware to capture it and convert it to a media format that RealServer can broadcast.

A third method, which uses on-demand clips but delivers them as if they were live, is available. It is not used as commonly.

- **Simulated live**-Just as a television broadcaster might record a live event and broadcast it later, such as Olympic sports that wouldn't be seen because of time zone differences, simulated live broadcasts take a pre-recorded event and broadcast it as a live event. Thus, although it is pre-recorded, users view the event as if it were live.

The table below summarizes the three participation types.

User Participation Comparisons

On-Demand (through Streaming)	Live Delivery (through Unicasting, Splitting, or Multicasting)	Simulated Live Delivery (through Unicasting, Splitting, or Multicasting)
Can access presentations any time.	Can only access presentations while they're in-progress.	Same as live delivery.
Files are stored on disk.	Presentations don't exist as files.	Same as on-demand delivery.
Presentations always begin streaming at the beginning of the file.	Everyone sees the same part of the presentation at the same time-late-comers join in the middle.	Same as live delivery.
User can fast-forward through the clip or pause it at any point.	User plays the clip all the way through.	Same as live delivery.
Similar to a videotape of past Olympic event highlights.	Similar to live television coverage of an Olympic event.	Similar to a previously recorded Olympic event, delayed on television because of time zone differences.

Which Delivery Method Is Right for Me?

Once you have determined how you want the user to experience the clip (as on-demand or live), you choose which delivery method you will configure RealServer to use.

- **On-demand**-the choice is simple: streaming is the only delivery method.
- **Live and simulated live**-there are three ways to deliver the clip: unicasting, splitting, and multicasting.

On-Demand Streaming

Pre-recorded clips are delivered through a method called streaming. A user who clicks a link to an on-demand clip watches it from the beginning. The user can fast-forward, rewind, or pause the clip. See Chapter 10, "Streaming On-Demand Presentations".

Live Event Broadcasting

Live clips can be delivered in several different ways. As the administrator, you will decide which method to use based on your network needs. A user who clicks a link to a live clip joins the live event in progress; fast-forward, rewind, and pause are not available because the event is happening in real time.

Live clips are broadcast as they are created. These clips don't exist as files, since they are created as the live event happens. Live content can be saved into files through the live archiving feature; the archived files become on-demand content and are handled as such.

Unicasting

This is the simplest and most popular method for live broadcasting. It requires little or no configuration. Refer to Chapter 11, "Unicasting Live Presentations".

Splitting

Splitting is a term to describe how one RealServer can share its streams with other RealServers. Clients connect to these other RealServers, called splitters, rather than to the main RealServer where the streams originate. Splitting reduces the load on the source RealServer, leaving it free to distribute other broadcasts. This method moves the broadcasts closer to clients, improving the quality of service for them. See Chapter 12, "Splitting Live Presentations".

Multicasting

Multicasting is a standardized method for connecting large numbers of users with presentations delivered over a network or the Internet. Consult Chapter 13, "Multicasting Live Presentations".

Simulated Live Event Broadcasting

The same delivery options are available as for live broadcasting: unicasting, splitting, and multicasting. The only difference is that the event has already been recorded, and no connection to a production tool or encoder is needed. The G2SLTA program, included with RealServer, sends the on-demand file to RealServer as if it were a live event. See "Creating a Live Source with G2SLTA".

Summary

The following table shows the user participation styles and the accompanying delivery methods.

Comparison of Delivery Methods

User Participation	Delivery Method	Appropriate Use	Requirements
On-demand	Streaming	Presentations that are limited to the number of licensed connections, CPU speed, amount of RAM, and available bandwidth of a single machine.	Requires sufficient bandwidth to handle the number of clients connecting.
Live and simulated live	Unicasting	Broadcasts that are limited to the number of licensed connections, CPU speed, amount of RAM, and available bandwidth of a single RealServer machine.	Requires sufficient bandwidth to handle the number of clients connecting.
	Splitting	Broadcasts that are limited to the number of licensed connections, CPU speed, amount of RAM, and available bandwidth of all RealServer machines.	Requires at least two RealServers. Must configure source RealServer and splitter RealServers.
	Multicasting	Broadcasts that will be viewed by unlimited users around the globe on a multicast-enabled network.	Requires a multicast-enabled network. Can be combined with splitting to cover a greater geographical region where networks are not multicast-enabled.

In some cases, you can use more than one live delivery method at once, to reach the maximum number of users while minimizing network bandwidth.

- The combination of splitting and multicasting is described in "Splitting and Multicasting".

- The combination of unicasting and multicasting is described in "Requiring Multicast Access Rather than Unicast" (for back-channel multicast) and "Using Unicast as a Backup Method" (for scalable multicast).

Linking to RealSystem Content

Links to media clips served by RealServer have several components that tell RealServer how to serve the clip and where to look for the clip.

Content creators will put most links into Web pages. A user looking at a content creator's site will click the link, and through the process described in "How RealServer Works", will receive the media.

For example, the following link for a RealVideo file would appear in a Web page (the URL for the media clip is shown in bold):

```
<a href="http://realserver.company.com:8080/ramgen/Concerts/French/
debussy.rm">Click here to watch today's concert!</a>
```

The clip may be pre-recorded, live, or pre-recorded but delivered as live.

Additional Information
Instructions on creating links to RealSystem clips are described in depth in Chapter 5, "Understanding Link Formats".

Working with Other Webcasting Professionals

This manual assumes that you (the RealServer administrator) are managing your RealServer, and that a second person (the content creator) is making media clips and SMIL presentations and putting links in Web pages. In reality, you may be filling both roles, especially when you are setting up a feature and want to do some quick tests. But it makes it easier to discuss the roles when they are described as separate people.

The RealServer administrator needs to provide the content creator with certain information, so that she can create the correct links in her SMIL files and Web pages. If the content providers are encoding live material, they will need to know where to direct their live data.

Responsibilities of RealServer Administrator and Content Creator

RealServer Administrator	Content Creator
Configures and maintains RealServer	Performs encoding or assembles presentations
Supplies information needed to create links	Creates links

Content Creators of On-Demand Content

Content creators will need the following information:

- Location where they should place their files
- Address or name of RealServer
- Port numbers for each protocol (but only if you have changed them from the recommended default settings)
- Information about whether Ramgen is in use (Ramgen is defined in "Ram Files and Ramgen" in Chapter 5, "Understanding Link Formats".

Content Creators of Live Content

In order to encode a live stream to RealServer, content creators need to know this information:

- Address or name of RealServer
- Port number to connect to
- Authentication information such as passwords (if any)
- URL to use in Web pages that point to a live broadcast or multicast
- URL to use in a SMIL file

Other RealServer Administrators

RealServer can broadcast to other RealServers, which can redistribute the presentations to clients, thus reducing the load on the original RealServer. This feature is called splitting. If you are working with the administrator of the other RealServer (the splitter), you will need to give that person certain information about your RealServer settings. That information is outlined in Chapter 12, "Splitting Live Presentations".

Firewall Administrators

If there are users within your network that either cannot receive presentations from RealServers on the Internet or who receive poor quality streams, information in Chapter 9, "Firewalls and RealServer" will help the firewall administrator understand what changes can be made that will enhance the users' experience.

Network Administrators

In determining both how much bandwidth is available on your network, and how much is appropriate for RealServer to use, network administrators can help you arrive at suitable numbers.

RealServer Features

In addition to the delivery methods described earlier in this chapter, RealServer has other features that help you administer your RealServer.

RealSystem Administrator

RealSystem Administrator is the Web-based console for customizing RealServer features. It can be run from any browser on your network. It is password-protected when first installed, and you can create additional user names and passwords for any other people who will be helping you administer your RealServer.

Access Control

The access control feature lets you associate certain client addresses with the ability or permissions to connect to certain ports.

Authentication

Authentication verifies the identity of a user or RealPlayer that is making a request for streamed media. The verification can come in the form of asking for a name and password, or it can be hidden from the user.

ISP Hosting

RealServer works with your existing user accounts and directory structure to make users' media files available for streaming. You allocate a minimum and maximum number of connections for each account, based on the number of streams permitted by your license. Allocating on a per-connection basis, rather than by stream, ensures that all files, including SMIL files which reference multiple streams, will always be served.

Monitoring

RealSystem Administrator includes a real-time Java Monitor to show activity on your RealServer, making Server management easy. It shows who is using the Server, when it is most used, and which files are the most requested, as well as other information.

Reporting (Log Files)

RealServer can create reports of historical data that let you see trends and gather information. Track who visited your site and for how long; what clips they watched and whether they watched them all the way through to completion. This information is stored in the access log. Any error messages are recorded in the error log. Requests for streams which will be cached are stored in the cached requests log.

Ad Streaming

RealServer can dynamically insert ads into streaming presentations. Offering integration with any HTML-based ad serving system, RealServer uses SMIL (Synchronized Multimedia Integration Language) to lay out ads and requested content in RealPlayer. This chapter explains how to set up RealServer's ad streaming features.

RealProxy

RealProxy is software that stores streamed media. While it is not part of RealServer, it can work with RealServer to share the distribution load, thereby conserving bandwidth over an intranet and allowing RealServers to send streams to a wider audience. It is generally installed on an intranet or on a large Internet Service Provider (ISP). When a client on the intranet or hosted by the ISP requests a streamed media file, RealProxy intercepts the request and sends it on behalf of the client. RealProxy then stores the requested media and streams it to any other clients who subsequently request the same material.

Firewalls

Firewalls are not specifically a RealServer feature, but they are important in networked environments. A firewall is a software program that monitors, and sometimes controls, all transmissions between an organization's internal network and the Internet. A network can consist of a company's local area networks, wide area networks, and the Internet, or it can be just an Internet Service Provider preventing inappropriate access to the files of its customers. The firewall's role is to ensure that all communication, in both directions, conforms to the organization's security policies.

Using RealServer Features Together

RealServer components can be combined to conserve bandwidth and deliver high-quality presentations. The table below summarizes RealServer features and how they work together.

For additional information on exactly how any of these features work together, refer to the chapter that describes the feature.

Interoperability of RealServer Features

	Streaming	Unicasting	Archiving	Simulated Live	Push Splitting	Pull Splitting	Back-Channel Multicasting	Scalable Multicasting	RealProxy Access	Firewalls[1]	Access Control	Authentication
On-Demand Delivery												
Streaming	·	-	-	-	-	-	-	-	·	·	·	·
Live Delivery												
Unicasting	-	·	·	·	-	-	§	§	-	·	·	·
Archiving	-	·	·	§	-	-	§	§	-	-	-	-
Simulated Live (G2SLTA)	-	·	§	·	§	§	§	§	-	-	·	·
Splitting-Push	-	-	-	§	·	§	§	§	-	·	·	·
Splitting-Pull	-	-	-	§	§	·	§	§	-	·	·	·
Multicasting-	-	§	§	§	§	§	·	-	-	·	·	·

Back-Channel												
Multicasting-Scalable	-	§	§	§	§	§	-	·	-	·	·	·
Other Features												
RealProxy Access	·	-	-	-	-	-	-	-	·	-	·	·
Firewalls[1]	·	·	-	·	·	·	·	·	-	·	·	·
Access Control	·	·	·	·	·	·	·	·	·	·	·	·
Authentication	·	·	§	·	·	·	·	·	·	·	·	·
ISP Hosting	·	-	-	-	-	-	-	-	·	·	·	-
Monitoring	·	·	-	·	·	·	·	-	-	-	·	·
Reporting (Log Files)	·	·	-	·	·	·	·	§	·	-	·	·
Ad Streaming	·	·	-	·	§	§	§	§	·	·	·	§

· Features work together automatically; no additional configuration required beyond normal setup

§ Requires some special considerations for these features to work together

- Not applicable; features are unrelated

[1] Firewall information assumes that firewall allows streaming media traffic and multicast traffic

"How Internet Radio Works," Howstuffworks,
http://computer.howstuffworks.com/internet-radio.htm/printable

home Search go

ComputerStuff AutoStuff ElectronicsStuff ScienceStuff HomeStuff EntertainmentStuff HealthStuff MoneyStuff TravelStuff PeopleStuff

Main > Computer > Internet

Click here to go back to the normal view!

How Internet Radio Works

by Debra Beller

A college student in Wisconsin listens to a disc jockey in Jamaica play the latest rapso (calypso rap) music. A children's advocacy group unites its geographically diverse members via private broadcast. A radio listener hears an ad for a computer printer and places an order immediately using the same medium on which he heard the ad. All of this is possible with Internet radio, the latest technological innovation in radio broadcasting since the business began in the early 1920s.

Internet radio has been around since the late 1990s. Traditional radio broadcasters have used the Internet to simulcast their programming. But, Internet radio is undergoing a revolution that will expand its reach from your desktop computer to access broadcasts anywhere, anytime, and expand its programming from traditional broadcasters to individuals, organizations and government.

In this edition of **HowStuffWorks**, we'll explore the Internet radio revolution in terms of equipment, transmission, programming and the alterations in the listener/broadcaster relationship.

Freedom of the Airwaves

Radio broadcasting began in the early '20s, but it wasn't until the introduction of the transistor radio in 1954 that radio became available in mobile situations. Internet radio is in much the same place. Until the 21st century, the only way to obtain radio broadcasts over the Internet was through your PC. That will soon change, as wireless connectivity will feed Internet broadcasts to car radios, PDAs and cell phones. The next generation of wireless devices will greatly expand the reach and convenience of Internet radio.

Uses and Advantages

Traditional radio station broadcasts are limited by two factors:

- the power of the station's transmitter (typically 100 miles)
- the available broadcast spectrum (you might get a couple of dozen radio stations locally)

Internet radio has **no geographic limitations**, so a broadcaster in Kuala Lumpor can be heard in Kansas on the Internet. The potential for Internet radio is as vast as cyberspace itself (for example, Live365 offers more than 30,000 Internet radio broadcasts).

In comparison to traditional radio, Internet radio is **not limited to audio**. An Internet radio broadcast can be accompanied by photos or graphics, text and links, as well as interactivity, such as message boards and chat rooms. This advancement allows a listener to do more than listen. In the example at the beginning of this article, a listener who hears an ad for a computer printer ordered that printer through a link on the Internet radio broadcast Web site. The relationship between advertisers and consumers becomes more interactive and intimate on Internet radio broadcasts. This expanded media capability could also be used in other ways. For example, with Internet radio, you could conduct training or education and provide links to documents and payment options. You could also have interactivity with the trainer or educator and other information on the Internet radio broadcast site.

Internet radio programming offers a **wide spectrum of broadcast genres**, particularly in music. Broadcast radio is increasingly controlled by smaller numbers of media conglomerates (such as Cox, Jefferson-Pilot and Bonneville). In some ways, this has led to more mainstreaming of the programming on broadcast radio, as stations often try to reach the largest possible audience in order to charge the highest possible rates to advertisers. Internet radio, on the other hand, offers the opportunity to expand the types of available programming. **The cost of getting "on the air" is less** for an Internet broadcaster (see the next section, "Creating an Internet Radio Station"), and Internet radio can **appeal to "micro-communities" of listeners focused on special music or interests**.

Creating an Internet Radio Station

What do you need to set up an Internet radio station?

- CD player
- Ripper software (copies audio tracks from a CD onto a computer's hard drive)
- Assorted recording and editing software
- Microphones
- Audio mixer
- Outboard audio gear (equalizer, compressor, etc.)
- Digital audio card
- Dedicated computer with encoder software
- Streaming media server

Getting audio over the Internet is pretty simple:

1. The audio enters the Internet broadcaster's encoding computer through a sound card.
2. The encoder system translates the audio from the sound card into streaming format. The encoder samples the incoming audio and compresses the information so it can be sent over the Internet.
3. The compressed audio is sent to the server, which has a high bandwidth connection to the Internet.
4. The server sends the audio data stream over the Internet to the player software or plug-in on the listener's computer. The plug-in translates the audio data stream from the server and translates it into the sound heard by the listener.

There are two ways to deliver audio over the Internet: downloads or streaming media. In **downloads,** an audio file is stored on the user's computer. Compressed formats like MP3 are the most popular form of audio downloads, but any type of audio file can be delivered through a Web or FTP site. **Streaming audio** is not stored, but only played. It is a continuous broadcast that works through three software packages: the encoder, the server and the player. The **encoder** converts audio content into a streaming format, the **server** makes it available over the Internet and the **player** retrieves the content. For a live broadcast, the encoder and streamer work together in real-time. An audio feed runs to the sound card of a computer running the encoder software at the broadcast location and the stream is uploaded to the streaming server. Since that requires a large amount of computing resources, the streaming server must be a dedicated server.

Lots More Information!

Related HowStuffWorks Articles

- How Radio Works
- How the Radio Spectrum Works
- How Satellite Radio Works
- How Ham Radio Works

More Great Links

- Radio-Locator: Internet Radio Search Engine
- SaveInternetRadio.org
- Kids Internet Radio
- RadioTower.com Internet Radio Directory

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"Telecommunications and Personal Management Services Linked in Collaboration by Verizon and Microsoft," October 23, 2001,http://www.microsoft.com/presspass/press/2001/oct01/10-23MSVerizonPr.asp

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Telecommunications and Personal Management Services Linked In Collaboration by Verizon and Microsoft

Key .NET Technologies to Help Verizon Customers Balance Family and Work Life

NEW YORK and REDMOND, Wash. — Oct. 23, 2001 — Balancing family, social and professional responsibilities can be overwhelming, but some innovative work by Verizon and Microsoft Corp. seeks to make the daily juggle much more manageable.

Microsoft and Verizon are exploring new uses of technology to integrate the latest telecommunications services, Verizon e-business applications and select Microsoft® .NET and Windows® XP services to provide customers with additional control over their lives. These technologies include telecommunications and messaging services, calendaring and personal directories. Features of .NET and Windows XP services now offered by Microsoft are playing an important role in one such application currently being tested by Verizon.

With a service bearing an internal code name Digital Companion, Verizon is working to extend the capabilities and features of its telecommunications services, already provided through one of the world's most advanced and pervasive networks.

"Being in a wired world should mean greater productivity and more control for people, and this is a key driver for our efforts," said Shaygan Kheradpir, president of eBusiness for Verizon. "Digital Companion would enable customers to access and use call management features, such as Caller ID, over the Internet, in new and innovative ways."

One version of the Digital Companion will use Microsoft's .NET Alerts to extend the reach of the service for Verizon's customers and will also use the .NET Passport authentication and single sign-in service to provide an easier, faster and more compelling experience.

"This effort is a great example of the kinds of customer relationships that are enabled by .NET," said Sanjay Parthasarathy, senior vice president of the .NET Strategy Group at Microsoft. "Verizon has combined its industry-leading telecommunications services with the smart clients, servers and services that make up the .NET platform to create a truly empowering communications experience for customers."

Anytime, Anywhere Communications

Based on the collaborative efforts of Verizon and Microsoft, this implementation of Digital Companion would provide a new way for people to more efficiently manage their day-to-day communication.

For instance, a Digital Companion user who is a working mother could get a Caller ID notification through an instant message popping up on her desktop computer signaling that her son's school has called her home. Without missing a beat, mom would

Related Links

- **Microsoft Professional Developers Conference Virtual Pressroom**

Transcript:

- Bill Gates Remarks - Oct. 23, 2001

Feature Stories:

- Thousands of Software Developers See .NET Technologies In Action - Oct. 23, 2001
- On Any Device: .NET Goes Mobile With the .NET Compact Framework - Oct. 23, 2001
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Press Releases:

- Microsoft Previews Global XML Web Services Architecture - Oct. 23, 2001
- Gates Rallies Developers For New Era of Computing - Oct. 23, 2001
- Massive Industry and Developer Support for Microsoft .NET on Display At Professional Developers Conference 2001 - Oct. 22, 2001
- Microsoft and Trans World Entertainment Announce Broad-based Business, Technology and Marketing Alliance To Deliver Enhanced Shopping Experience For FYE Customers - Oct. 22, 2001

Microsoft Resource:

- Microsoft .NET Web Site

open her Digital Companion and find out that the school has left her a voice message.

By listening to her voice mail, mom would learn that her son is ill and needs to be picked up early. Rather than digging through her address book to find the number of the school, mom scans her caller ID list in her Digital Companion and calls the school to ask if her son requires immediate medical attention.

The school tells her not to worry; her son will be fine, but needs to be picked up early from school. Since critical business calls are expected that afternoon, mom could return to her Digital Companion and forward all her calls to her cell phone just before she leaves to get her son.

Finally, mom could prepare for spending the next day at home and the doctor's office by directing certain important calls to her cell phone and others to her home office. Her family commitments are met, and she never misses a beat with her work.

Utilizing key .NET technologies, Digital Companion would enable a user to remotely access features of Verizon's existing call management services, such as Caller ID and voice mail, any time, anywhere and from virtually any device. With Digital Companion, customers would no longer have to check in at home or work when traveling. Instead, using the remote access to call forwarding provided through Digital Companion, calls could be routed to a cell phone, hotel room or temporary office for the duration of the trip, providing an unprecedented level of convenience for participating customers. Verizon Caller ID lists could also be checked remotely for the first time using this service. Verizon plans to conduct technical trials of Digital Companion in the near future.

Additional Collaboration

While Verizon and Microsoft have worked together in the past, the collaboration around the Digital Companion project is a unique example of the companies' focus to create new services that transform the customer experience. This initiative uses the technical advancements from Microsoft and Verizon, by integrating e-business, telecommunications and software infrastructure to build products that can form the basis for a new generation of communication experience.

Verizon uses Windows 2000 and Microsoft SQL Server (TM) 2000 in several of its key customer relationship management systems. These technologies have enabled Verizon to further enhance the customer experience and resulted in development productivity and improved system performance.

In addition, Verizon and Microsoft are working together to deliver cutting-edge services to Web users by making Verizon's SuperPages.com directory services available on the MSN® network of Internet services. And, recently, Verizon and Microsoft announced an agreement for Verizon to provide broadband digital subscriber line (DSL) access to MSN Internet Access customers.

About Verizon

Verizon Communications (NYSE:VZ) is one of the world's leading providers of communications services. Verizon companies are the largest providers of wireline and wireless communications in the United States, with 125 million access line equivalents and more than

28 million wireless customers. Verizon is also the world's largest provider of print and online directory information in the world. A Fortune 10 company with about 260,000 employees and more than $65 billion in annual revenues, Verizon's global presence

extends to 40 countries in the Americas, Europe, Asia and the Pacific. More information on Verizon can be found at http://www.verizon.com/.

About Microsoft

Founded in 1975, Microsoft (Nasdaq "MSFT") is the worldwide leader in software, services and Internet technologies for personal and business computing. The company offers a wide range of products and services designed to empower people through great software -- any time, any place and on any device.

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Note to editors: If you are interested in viewing additional information on Microsoft, please visit the Microsoft Web page at http://www.microsoft.com/presspass/ on Microsoft's corporate information pages.

“Real-Time Collaboration Integration in the Portal,” T. Odenwald, SAP Design Guild,http://www.sapdesignguild.org/editions/edition5/synch_collab.asp

print version of the article

Real-Time Collaboration Integration in the Portal

By Thomas Odenwald, SAP Labs, Palo Alto

An article in the *Financial Times* of July 12, 2002 stated: "Bill Gates had complained this year that his documents, e-mail, and instant messaging buddy list did not work together and were not related to his calendar."

We could certainly help Mr. Gates with our new solution for the *SAP Collaboration Room* (Enterprise Portal Edition). Not only do we enable single point of access for document management within teams and projects, provide synchronization with the users groupware solutions, and offer unified calendar functionality, we also enable real-time collaboration integration through our own integrated solution, as well as through integration of third-party service providers.

What is Real-Time Collaboration?

The classical real-time collaboration scenarios are:

- Desktop sharing
- Application sharing
  Share applications, documents, or desktop to enable online meetings, remote support, and so on.
- Co-browsing
  Share browser (see application sharing)
- Buddy list and awareness
  A user's ability to maintain a list of people that are the ones he/she usually interacts with and to be able to see which of them is currently online, offline, busy, and so on.
- Whiteboard collaboration
  The electronic equivalent of a blackboard and chalk, but between remote users. Whiteboard systems allow network participants to simultaneously view one or more users writing on an on-screen blackboard or running an application.
- Instant messaging (IM)
  The ability to exchange immediate messages with connected buddies
- Chat service

Groupware and Obtaining User Acceptance
- Supporting Groupware with Awareness Information
- Personal Networks
- Supporting Group Work through Hardware and Software Solutions

SAP Collaboration Projects
- SAP Collaboration Room (mySAP Enterprise Portal Edition)
- Real-Time Collaboration Integration in the Portal

Legend

Book Document

An enhancement to IM where multiple users can exchange information while still being able to see previous messages

Additional features have emerged recently like

- Voice over IP (two-way audio transmission)
- Video and audio conferencing capabilities
- Annotation tools

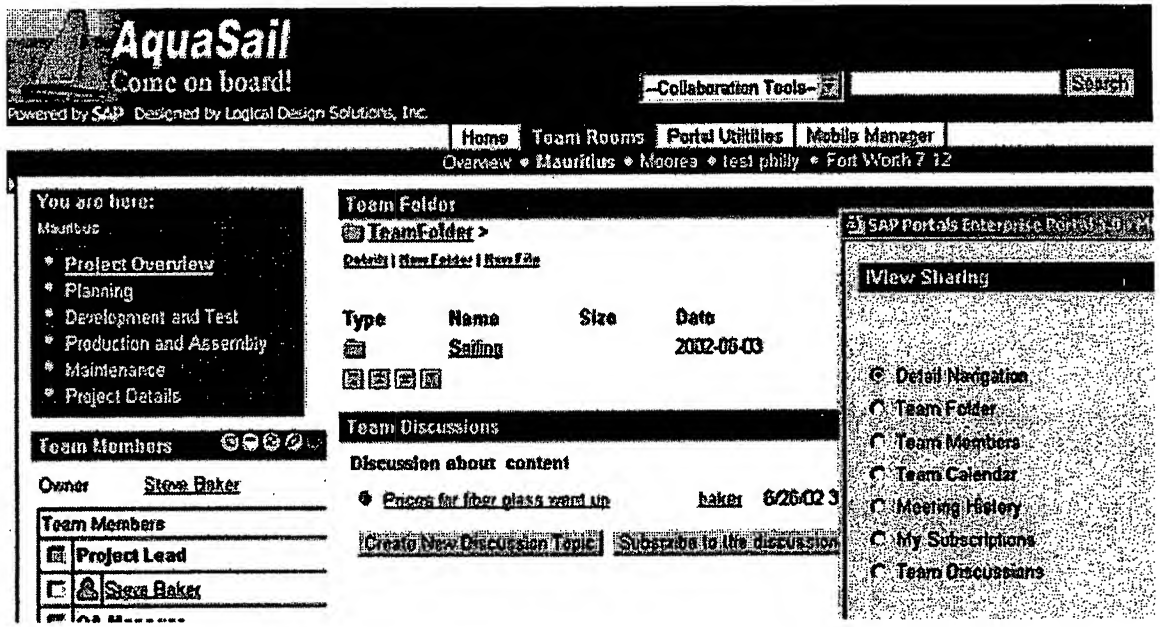
Recent terrorist threats and the economic situation are encouraging many companies to take a closer look at these features to avoid unnecessary travel, decrease overall expenses and face-to-face time, lower cost of ownership, and increase productivity. Increasing bandwidth and cheaper hardware makes these tools more attractive by the day.

Gartner Group even predicts that *instant messaging* will overtake classical e-mail in the near future (cf. magazine 15/01).

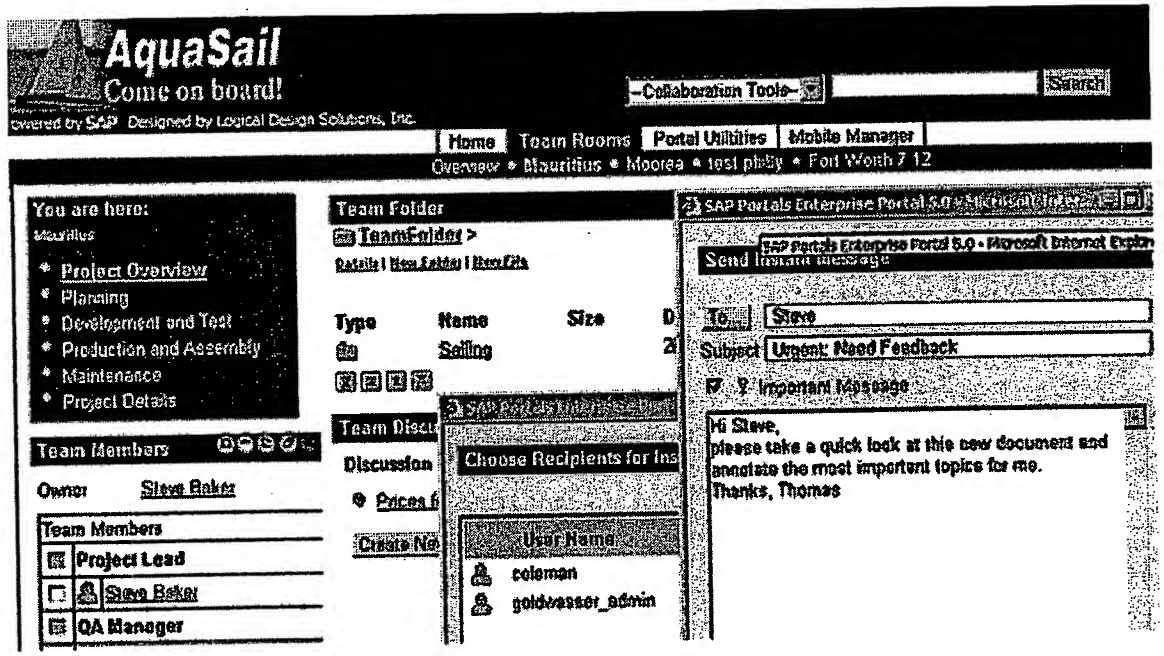
The tremendous value of Web-based real-time interactivity is not in doubt, especially where responses are needed immediately, consensus time needs to be reduced, and the decision makers are distributed around the globe.

There are lots of options that enable both individuals and teams in different locations to communicate in real time. However, because many of these options are not integrated into the familiar tools used in the daily work environment, they are not part of a cost-effective solution. The *SAP Collaboration Room* solution offers these real-time services through the portal and enhances the value of the information stored and accessed by sharing it in a contextual, ongoing team environment. Teams and individuals are provided with a tool set to enable them to collaborate more easily and efficiently.

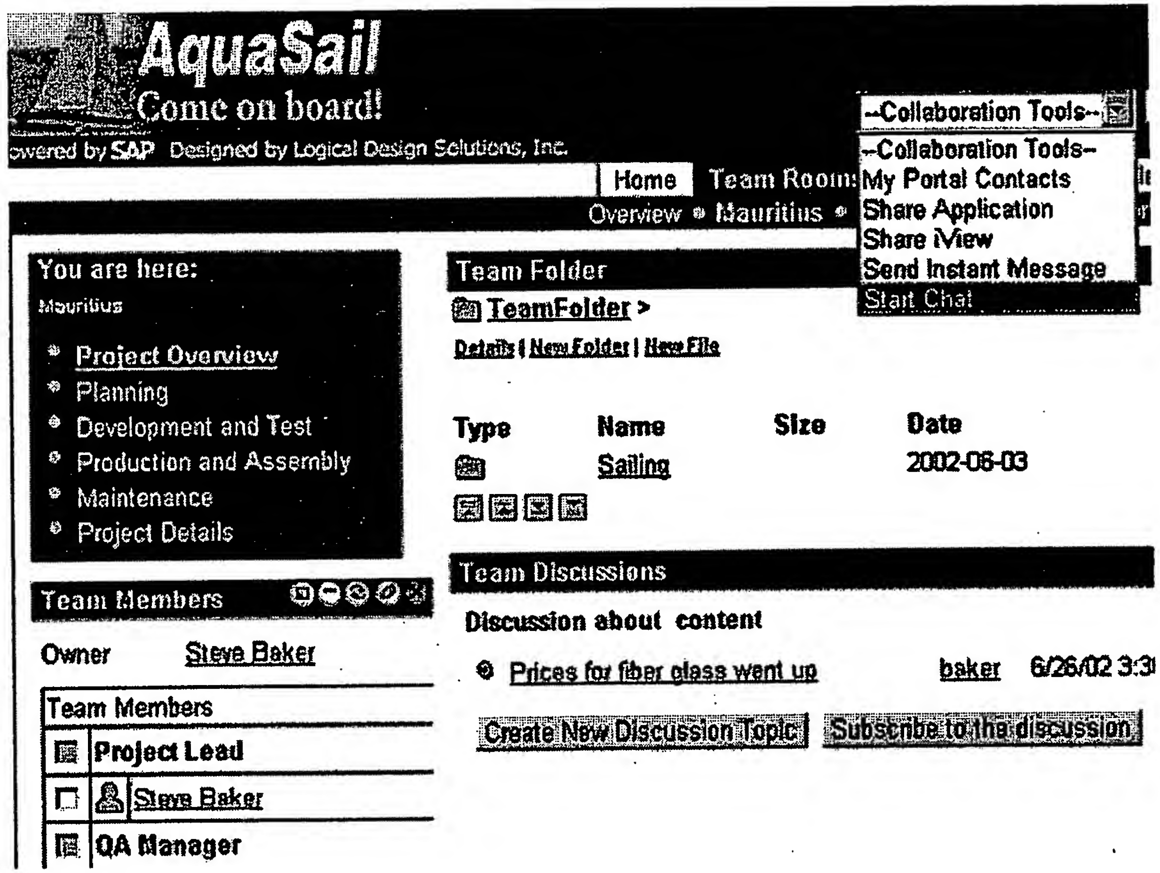
Examples:
- Share iViews or applications from any portal page instantly



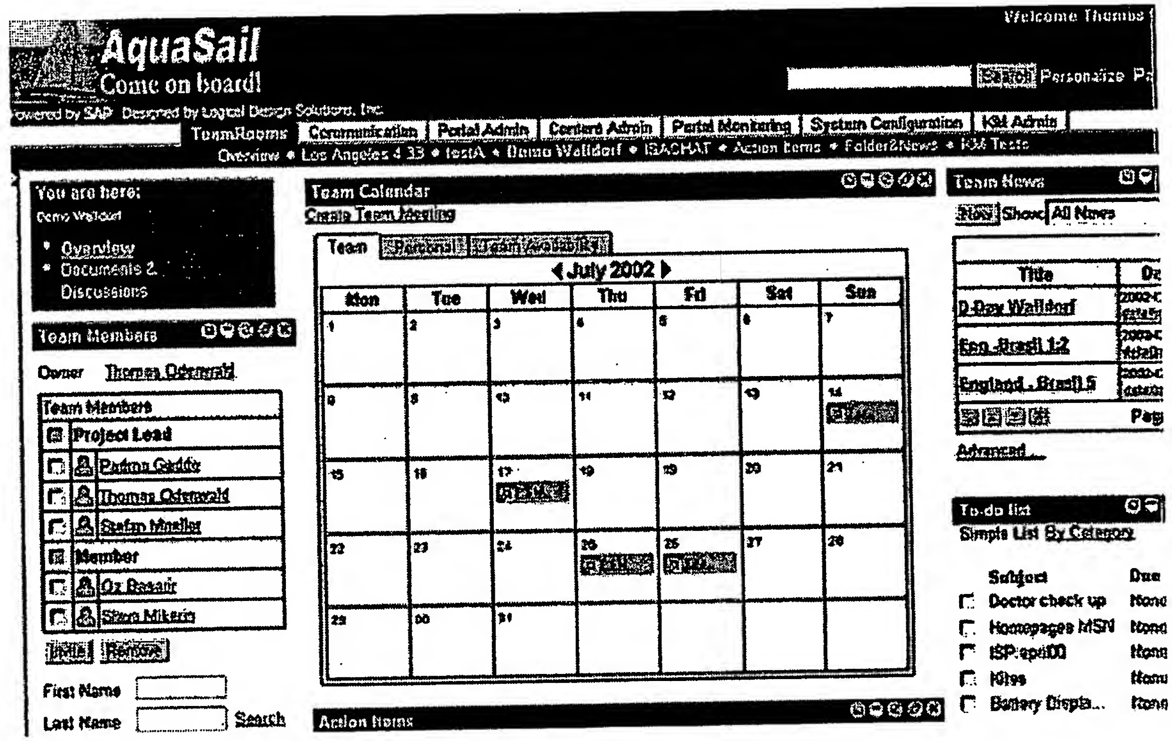
- Create instant message instantly from the portal by always knowing who is logged on to the portal and who is currently unavailable

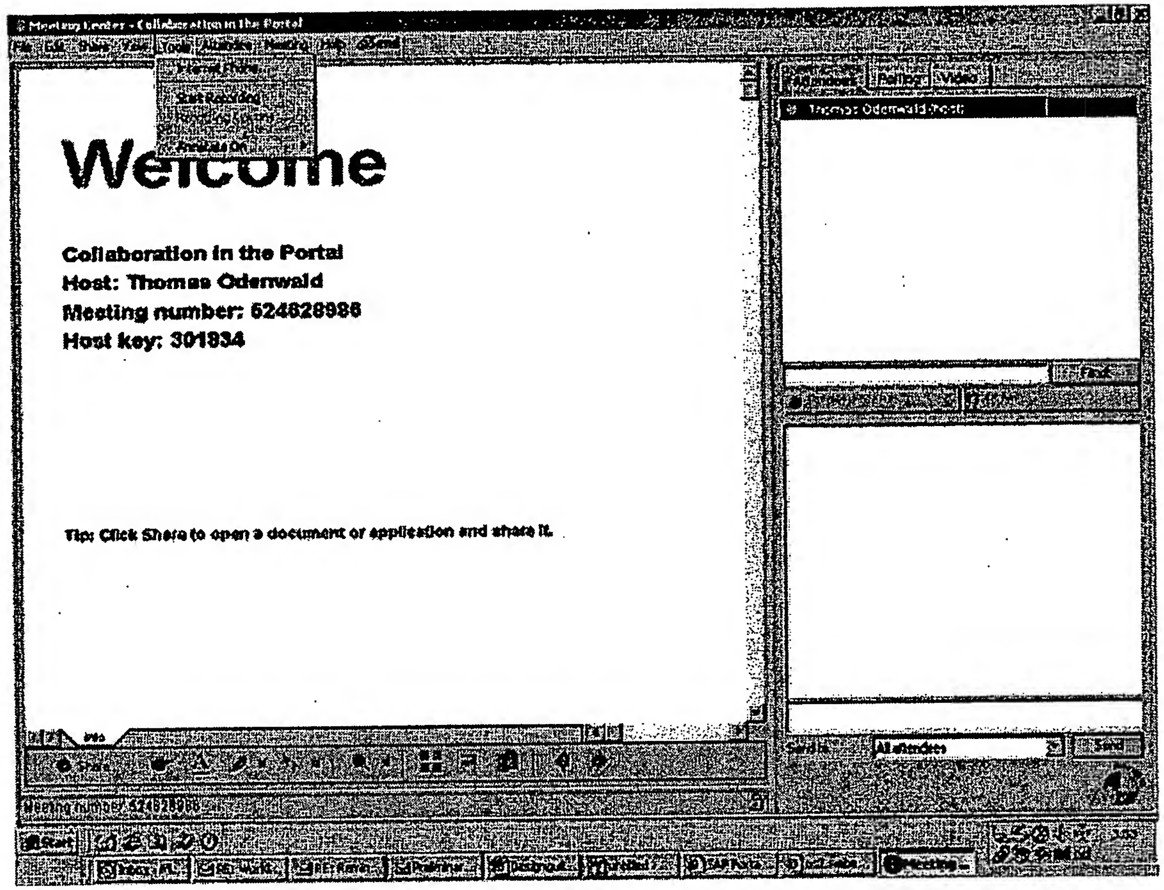


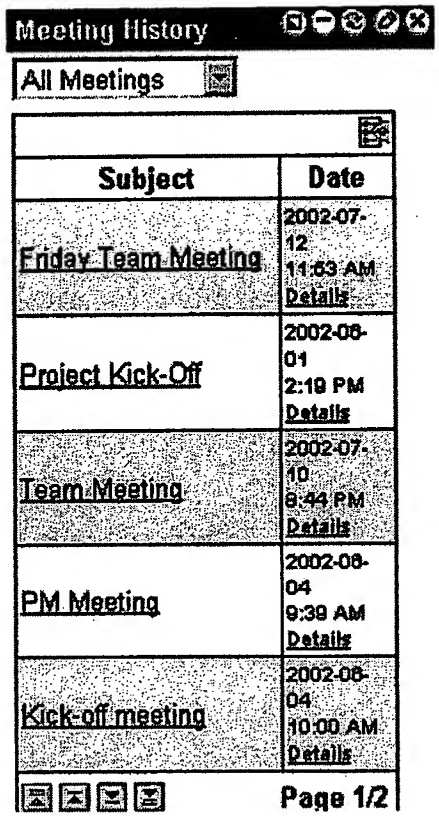
- Start public and private chats within the portal



- Schedule online team meeting, start meeting with mouse-click, and see the meeting results instantly in meeting history


AquaSail
Come on board!
Welcome Thomas
Personalize
TeamRooms
Communication
Portal Admin
Content Admin
Portal Monitoring
System Configuration
KM Admin
You are here:
Overview
Team Calendar
Create Team Meeting
July 2002
Mon
Tue
Wed
Thu
Fri
Sat
Sun
Team News
Title
D-Day Walldorf
Team Members
Owner
Project Lead
Member
First Name
Last Name
Search
Action Items
To-do list
Subject
Due
Doctor check up
Homepages MSN
Kites
Advanced...


Welcome
Collaboration in the Portal
Host: Thomas Odenwald
Meeting number: 524828986
Host key: 301834
Tip: Click Share to open a document or application and share it.



In this scenario valuable meeting information can be captured seamlessly by attaching any file or link to the automatically generated meeting record. Any audio and video recordings included can also be started with a mouse-click. In a sense, these options allow you to build up your own e-learning center.

Examples:

1. Integration of Yahoo! broadcasting feature
2. Integration of WebEx Player

And this last example might just solve Bill's problems:

- All project related documents are always available in form of iViews
- All e-mails, calendars, and tasks are available in form of iViews
- Instant messaging buddy list is available
- Project/team-related and personal calendar is unified and available

Value Proposition?

Success depends on collaborative teamwork — and real-time services meet the challenges of a distributed work force. With teams no longer based in a single location, collaboration cannot happen exclusively in face-to-face meetings or in coffee corners.

Companies are already saving millions of dollars by using real-time services. The result is a reduced information exchange time, with

- single point of access to relevant information
- closer collaboration
- easier knowledge sharing
- reduced travel and reduced travel costs

Summary

Real-time services on the portal can be used as an extension to the *SAP Collaboration Room* solution or as a stand-alone solution. Real-time collaboration enables collaboration between individuals and teams, and manages the content they create. Rather than replacing existing tools, it integrates and unifies existing solutions in the portal environment.

Information stored in different places is available through one point of access. Users know exactly where they should be storing, sharing, and collaborating on documents and can meet and exchange information in a real-time environment.

top

print version of the article

"NetMeeting101,"
http://www.meetingbywire.com/NetMeeting101.htmz

NetMeeting 101

Home
Site Map
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NetMeeting 102
NetMeeting List
MSN Messenger
H.323 Corner
XP Messenger
XP/NetMeeting
Tips and Stuff
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Gateways
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What is NetMeeting?

NetMeeting is a real-time communications tool from Microsoft that allows individuals to communicate in pairs or groups over the internet or intranet (an IP enabled LAN) using audio, video and data communication.

Why is NetMeeting different?

NetMeeting has a number of characteristics that make it better than other similar tools:

- it is free (downloadable at Microsoft - Win2000 and XP versions are preinstalled and will not allow installation of downloadable versions)
- it is standards based (which means that it can communicate with other standards based products)
- it operates with 2 or more individuals in a meeting
- it has built in audio, video, whiteboard, chat, file transfer, program sharing and collaboration functions

What is required to run NetMeeting?

- Windows 95/98, Me, NT, 2000 or XP
- Internet or Intranet connection (TCP/IP)
- A sound card with a microphone and speakers (or better yet a headset with integrated microphone) is desirable

What is not required to run NetMeeting?

- NetMeeting 3.0 requires IE4.01+ to be on the system -- though there is no requirement to use it as your default browser
- NetMeeting does not require a video camera to view callers (to send video a camera is required -- parallel port or USB cameras are the easiest to install but capture card based cameras work best - a review of cameras and guide to selection is on the hardware page)

What are directory or ILS servers?

Most users of the internet are on dial up lines - not connected all the time. Each time they connect their IP addresses change (the IP address is at the heart of the internet -- all location information ultimately gets translated to IP address and computers intercommunicate using that as an address).

In order for potential calling computers to connect to you, they must have two pieces of information:

- they must know that you are online or connected
- they must know your current IP address

The ILS servers supply the function of providing this information. When you "Log on" to an ILS server you are telling the ILS server that you (identified by your supplied email address) are

connected to the internet, that you are at a certain IP address (a current bug in NetMeeting actually records the incorrect IP address if you are on an IP enabled LAN but connected through a dial up line) and that you are running NetMeeting and able to field calls. Now potential callers need know only two fixed pieces of information to call you:

- the ILS server that you are logged into or would log into if you are online
- the supplied email address

The ILS server you choose to use need bear no relationship geographically to where you are -- all that is necessary is that your potential callers know which you will log in to.

The ILS server knows whether or not you are online and if you are what your current IP address is. You can "log in" to an ILS server without appearing in the viewable directory so that only associates that know the ILS server and email address you supplied can find you.

Changed January 2,2003

I can't log on to a server!

There could be a number of reasons for this problem:

- you have an AOL installation that installed a 16 bit Winsock -- if this is the case you cannot log on to any server
- you are behind a proxy server that is not allowing your connections to the ILS server
- you have just logged out and the logout is not yet completed
- somebody else is logged on using your chosen email address -- this might be especially true if you use a common pseudonym email address
- the server you want to user is busy or overloaded
- apparently Win2000 somehow dictates a different LDAP port number -- it is necessary to add the port number(389) to the ILS name (i.e. ils.xxx.com:389)

Changed April 1,2000

Microsoft has outlined many of these issues in the support database

I get logged out every few minutes!

It seems that NetMeeting, when used on a dial up line, depends on the ICMP protocol (the same one that is used by ping and traceroute programs) to assure a connection before it does its regular update to the ILS server (every 2 minutes apparently).

If your router, your ISP, the ISP providing the ILS server or someone between is blocking or dropping ICMP packets (this is common in busy internet situations -- the "non-essential" ICMP packets are the first to be dropped by busy routers) you may be logged out.

Changed February 28,1999

Microsoft has published a support article on this issue.

What happened to the Microsoft ILS Servers and Internet Directory?

On December 15,1999 Microsoft decided to withdraw all ILS servers that it was running and let 3rd party servers handle the load and instead concentrate on MSN Messenger as vehicle for initiating NetMeeting calls.

No support for ILS servers was changed in NetMeeting so if you wish to continue to use an ILS

server you should pick one of the 3rd party servers available. Use NetMeeting HQ to find servers.

Changed December 27,1999

Finding an ILS server

Since Microsoft closed their servers it has become difficult for many people to find usable servers.

A number of websites I have noticed are maintaining ILS server scanners -- allowing you to determine current state of servers and to search a number servers at once.

Changed April 30,2003

Check out the sites -- NetMeeting Headquarters and VideoFrog, NetMeetingserver.net

I am not listed what is the problem?

There are a number of reasons why you might not see yourself listed:

1. You are not logged in to an ILS server (logging in can happen automatically on NetMeeting start up or on command depending on your preferences set up)
2. You have indicated in options that you want your listing not to be visible (this is like an unlisted telephone number -- people can still call you but they must know your number)
3. You are looking at a different ILS than the one you logged into.
4. You are looking at different category of user than the category you indicated on your login.
5. The ILS is overloaded or has failed
6. You have not refreshed the list to the most current available.

Changed June 9,2000

Why do I never get calls or why can't people who try to call reach me?

NetMeeting ILS registration works by registering your current IP address in the ILS system. People can call you through the ILS only if the address registered in the ILS is correct and reachable.

Various circumstances cause this not to be true:

- You are running on a TCP/IP enabled LAN and use a dial up ISP.
- You are running a TCP/IP enabled LAN and go through a proxy, firewall or NAT translator.
- You are running a TCP/IP enabled LAN and also use an Ethernet card that connects to a cable modem.
- You have WebTV software installed

This problem was prevalent with V2.x. Fortunately in V3.0 this is no longer a problem - NetMeeting uses the IP address of the adaptor that it will be talking to the ILS on as the IP to register - so most of the previous aberrant behaviour has been removed.

Changed April 9,2002

Manage which IP you register in the ILS?

A suggestion from Cuseeme users (who apparently experience a similar problem) was that the IP used is the IP of the last installed adapter -- and that deinstallation of all adapters Ethernet and dial up) and reinstallation with the IP that you desired to be registered on the last installed adapter was a solution. If you try this make sure you record all the settings on the adapters

before you deinstall.

Changed February 16,1999

Another suggestion from a NetMeeting user is to deinstall NetMeeting, connect to the internet

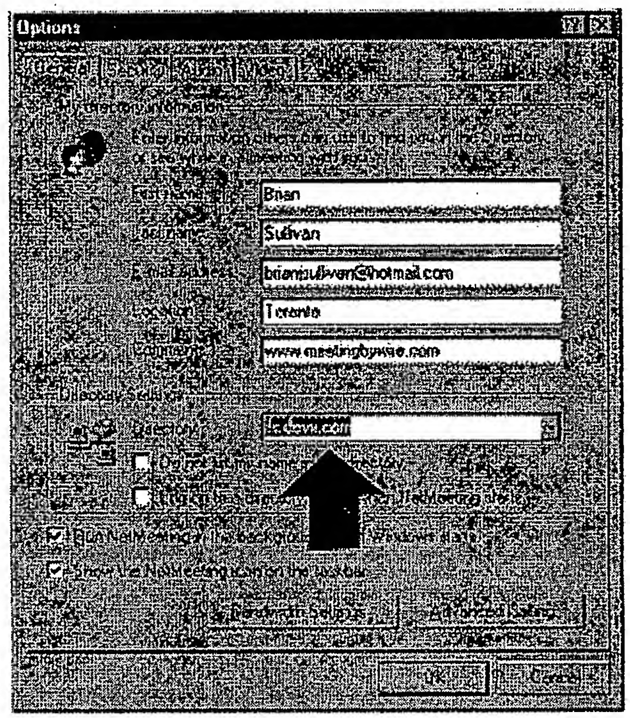
ILS theory vs. practice

Unfortunately the current ILS strategy is faulty and the overload problem mentioned above is becoming more and more common. In response to problems Microsoft has chosen to take all of its ILS servers permanently out of service and is favouring MSN Messenger as the prime connection strategy..

The ILS system itself seems unscalable and prone to breakdown or other faults. Nobody manages the system so a particular ILS server can go for days without functioning.

I have picked a new ILS server - how do I add it to the list?

To add an ILS server to the list of servers that you can log into or pick in the directory listing window click on the Server or Server name pulldown in the directory listing view or in the Tools/Options menu General tab (the server name text should now be selected) -- type the new name or paste the name in.



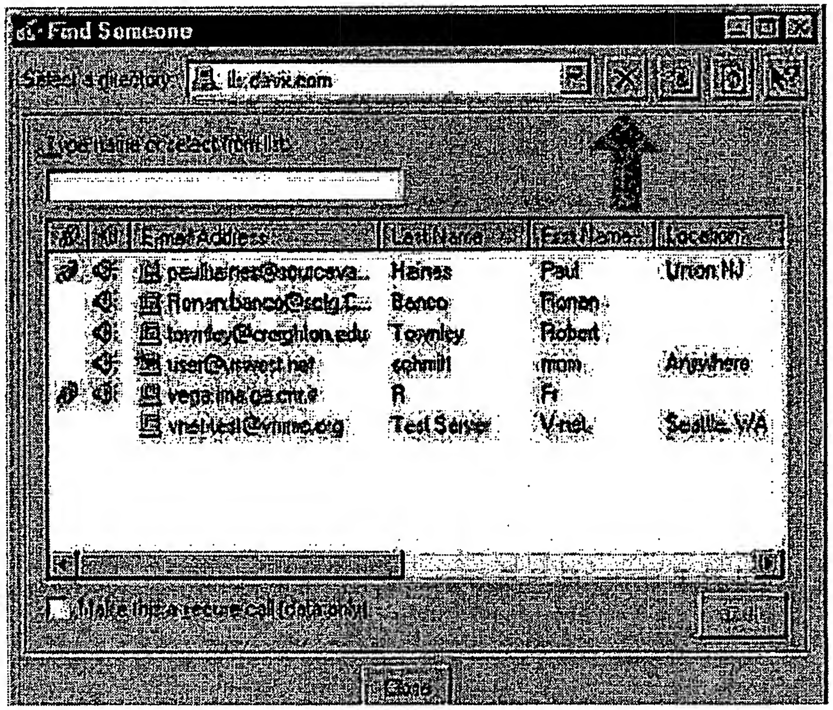
Changed July 22,1999

Can I delete

There is a limit of 15 servers allowed in the list. A function for deleting existing servers was

servers as well?

left out of the V 2.x of NetMeeting (to delete a server use the NetMeeting Super Enhancer program downloadable at the NetMeeting Place). To delete a server in 3.0 use Call menu. Directory item to display the "Find Someone" dialog. The current server can be deleted using the "X" button:



Added July 22,1999

Is there relief from this ILS dilemma?

There are a number of ways to avoid this ILS problem:

1. Find another less used more stable ILS server to register on. NetMeetingHQ and Videofrog , Chatpal also have ils server lists.
2. NetMeetingHQ and Videofrog have online ils scanners that provide web lists of ils servers. neetingserver.net has a similar service that is more family or work place friendly.
3. Use ICQ, AOL Instant Messenger or another buddy program as online signal and NetMeeting connection tool.
4. Use the TZO naming system , Dyndns or another dynamic naming service. This includes a permanent name that points your dynamic address.
5. Email partners informing them that you are online -- include your IP address (available by running ***winipcfg*** from the Start menu - run item for V2 and in the About box in V3) allowing them to call you directly.

Changed April 30,2003

6. Use Microsoft's MSN Messenger which has built in NetMeeting integration.

Making a Call using MSN/Windows Messenger

Since Microsoft closed all its ILS servers they have promoted Msn Messenger as the method for starting NetMeeting calls. Messenger is a much different mechanism than an ILS - though essentially it performs the same function as far as NetMeeting is concerned. It requires that you have a predetermined "buddy" list -- a list of individuals that have allowed you to determine their online status and to contact them.

Assuming you have downloaded and installed Msn Messenger and have established a list of buddies you can start a NetMeeting call from Msn Messenger using the "Start NetMeeting" menu item.

Unfortunately Microsoft in its infinite wisdom has chosen not to include this item in the XP "Windows Messenger" version. You can make it come back by setting the Messenger to Windows 2000 compatibility mode (right click on a Messenger shortcut, select properties, compatibility). For more information see the XP/NetMeeting page.

Changed January 02,2003

Can I avoid the ILS and the internet and dial directly?

Versions of NetMeeting prior to V2.1 had a feature where you could make direct calls (modem to modem) avoiding the Internet and the ILS. This feature was removed in V2.1 and later.

It can still be done however. Noël Danjou has published information on how to to do this (a Word document) available from his download page.

I haven't tried the instructions myself the information but they seem complete and detailed.

Changed August25,1999

Microsoft has published information as well.

Can I set up NetMeeting to run on a LAN with no Internet access?

To set up a functional intranet to use NetMeeting:

All computers must have the TCP/IP protocol installed and bound to the Ethernet card and assigned TCP/IP addresses -- there are a number of ways to do this but the easiest is to assign an IP address from the non routable private set (I use 192.168.0.x with a mask of 255.255.255.0).

Disable Wins, and DNS unless machines have access to the internet or an internal DNS server. Install NetMeeting and set it up on all machines so that it: (all of these are in the Tools.. Options -- Calling tab) doesn't log on at start up.

Computers can now call each other in NetMeeting using the IP address 192.168.0.x as the address to call or computer name (identification tab of of Network properties).

There are other possible set-up strategies (IP addresses could be assigned by DHCP, you could run a DNS or WINS server to get computer name, you could run an internal ILS) but all add

extra complication.

Changed December 19,1999

Extensive LANs might require implementation of an ILS server to provide both presence and location information

Deleting the Most Recently used drop down list

The most recently used address drop down list (the black box above the video window) contains a list of the most recently typed in addresses to which completed calls where made.

It is stored in the registry at:
HKEY_CURRENT_USER\Software\Microsoft\Conferencing\UI\CallMRU

Deleting the key or contents will allow you to start the list anew -- I don't know the exact format of all entries but with some testing and manipulation you could probably add or remove individual items as well.

Added May 24,2001

Creating a Speedial list

Creating a series of .cnf files in the "speeddial" sub directory of the NetMeeting install directory will create an address book the can be reused and transferred to other machines. The directory is accessible via the Call -> Directory pane drop down list.

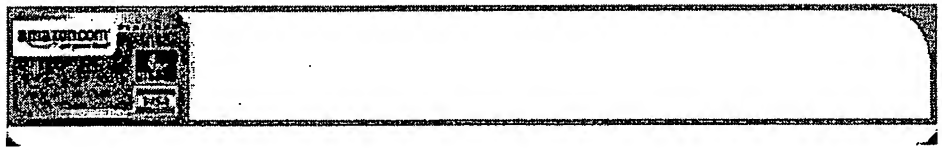
Added May 24,2001

Test your audio video setup

Added January 02,2003

Intel it seems is providing a test site to test your H.323 calling program (I think their own but it seems to work for NetMeeting.

Problems, comments, questions: Email webmaster
Changed:Monday October 06, 2003 12:28 -0400


amazon.com

"NetMeeting102,"
http://www.meetingbywire.com/NetMeeting102.htm

NetMeeting 102

Home
Site Map
NetMeeting 101
NetMeeting 102
NetMeeting List
MSN Messenger
H.323 Corner
XP Messenger
XP NetMeeting
Tips and Stuff
NetMeeting 3.0
Gateways
Contact Center
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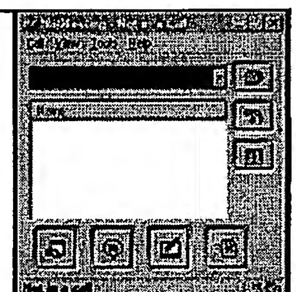
This page has a large number of graphics and may take a long time to load. I apologize but I think most of the information is best expressed with pictures - I hope it is worth the wait.

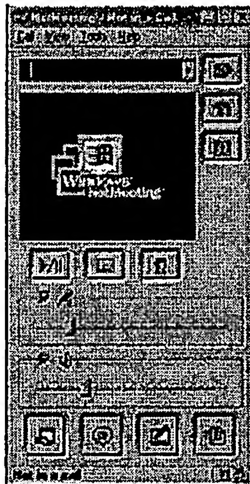
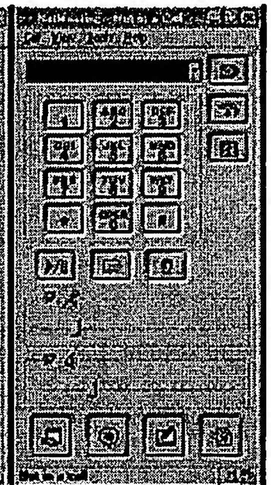
NetMeeting can be a useful tool but it can also be very frustrating. I have found that people sometimes get into difficulty after playing with various options and can't easily find their way to fix them. They end up with a program that is configured for some use that they never intended and can't make the kind of calls they want.

This page will list the most common problems I see with comments and solutions.

Incorrect View Selection (no video)

If your view of NetMeeting when you start up is not correct for video calls (no video window visible):

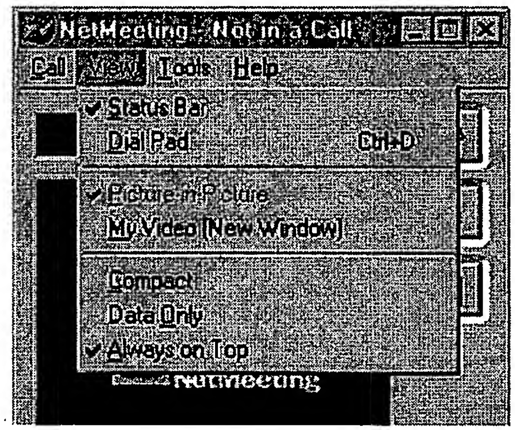


Correct	Incorrect	Incorrect

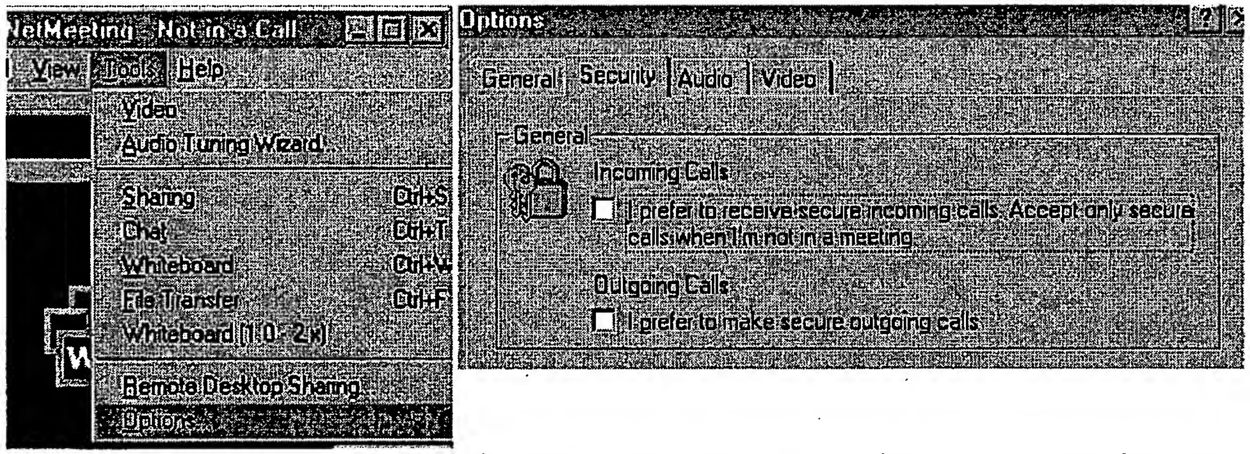
You have set your view so that you have no video window - this is a valid setting for some situations (using data only or use with a NetMeeting to telephone gateway) but for most people wanting to make audio/video calls these settings are incorrect.

To fix this problem make sure that the ***View*** menu does not have either ***Dial Pad*** or ***Data Only*** selected.



Incorrect Security Settings (no audio/video) - your settings incorrect

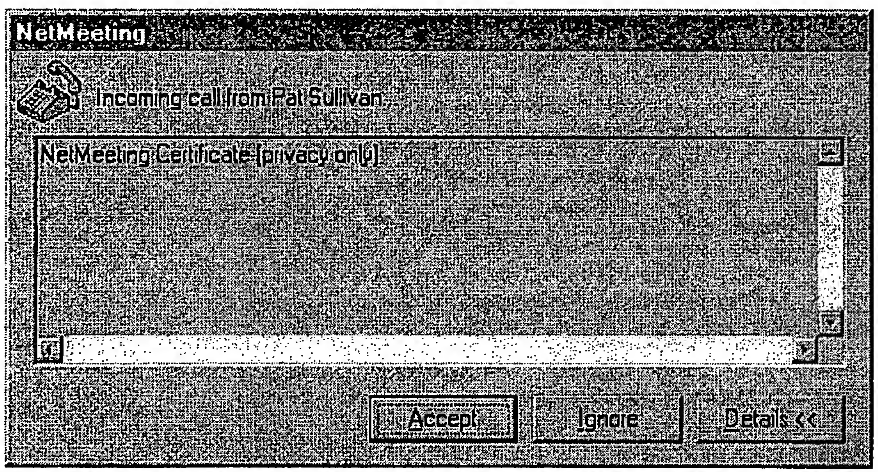
If you cannot connect to another party in an audio video call it may be because you or the other party have security settings that allow data only calls. In the ***Tools*** menu, ***Options*** item ***Security*** tab:



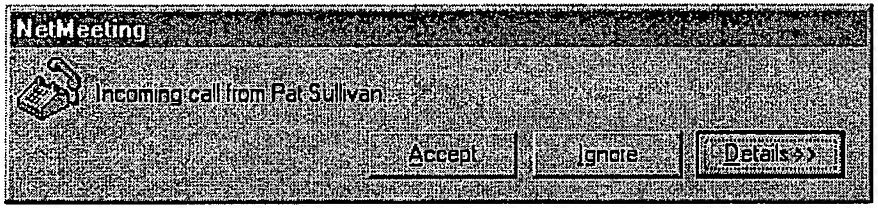
Make sure that the settings in the ***Security*** section about requiring ***Incoming Calls*** and ***Outgoing Calls*** to be secure are unchecked - making or receiving a "secure" call implies that it will be a data only call.

Incorrect Security Settings (no audio/video) - the other party's settings incorrect

If you receive an incoming call with a notice that looks like this:

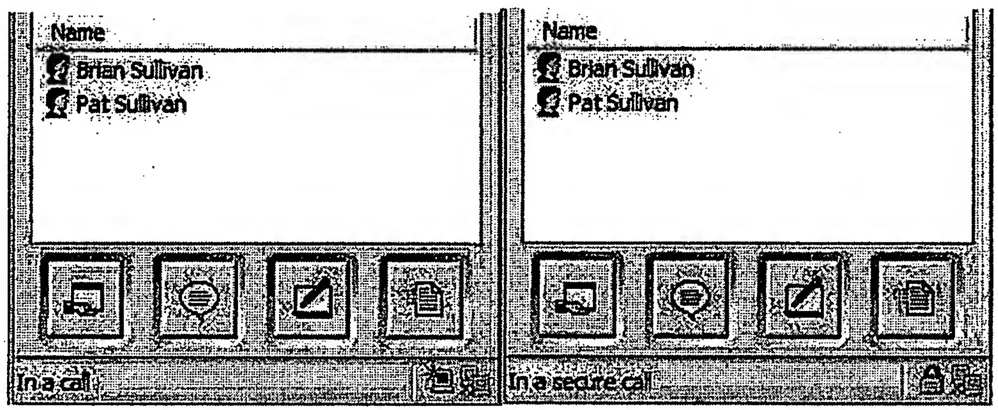


or this:



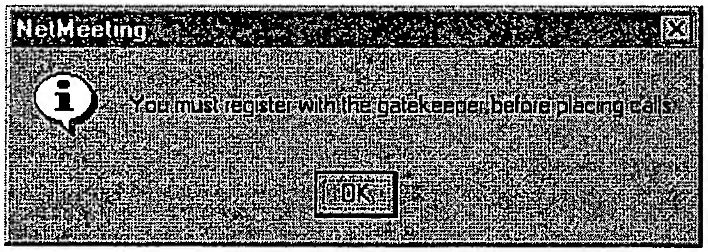
the other party has made a secure call which will be data only. This is what a normal incoming call looks like:

The two captures that follow show the difference between a secure call and a normal call in progress (notice the status bar at the bottom of the window).

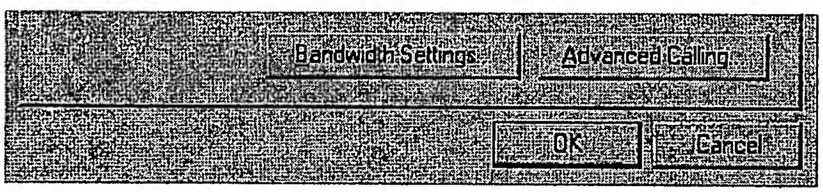


Gateway Configured but not Required

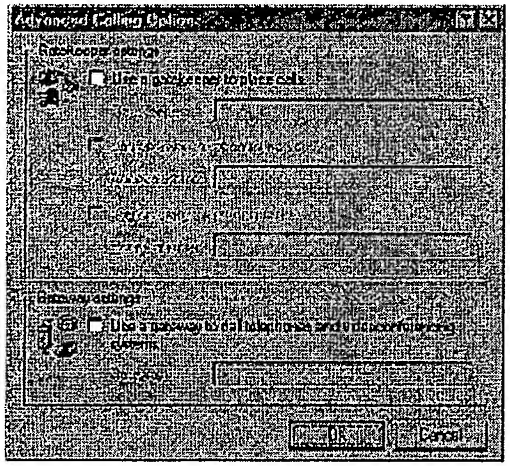
If you try make a call and get a message like:



Likely you have configured the ***Advanced Calling*** features incorrectly in the ***Tools*** menu ***Options*** item - ***General*** tab

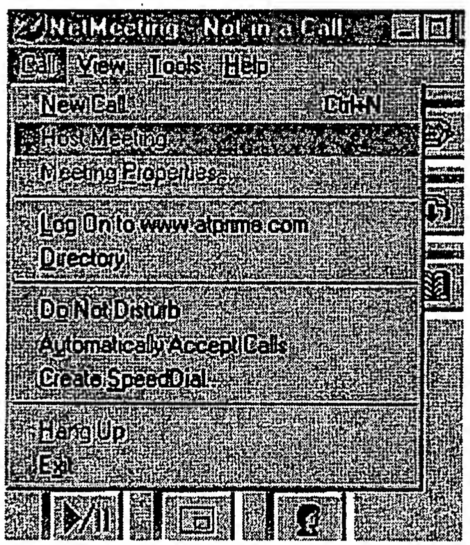


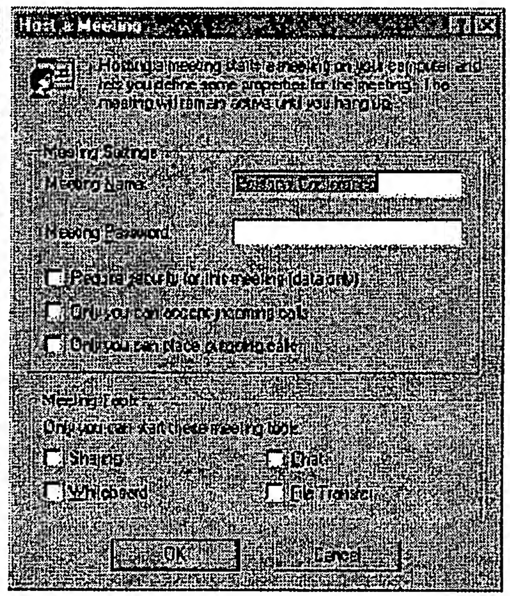
Clicking on the ***Advanced Calling*** shows dialog box like::



All items in this menu should be unchecked (as in the graphic) for normal operation.

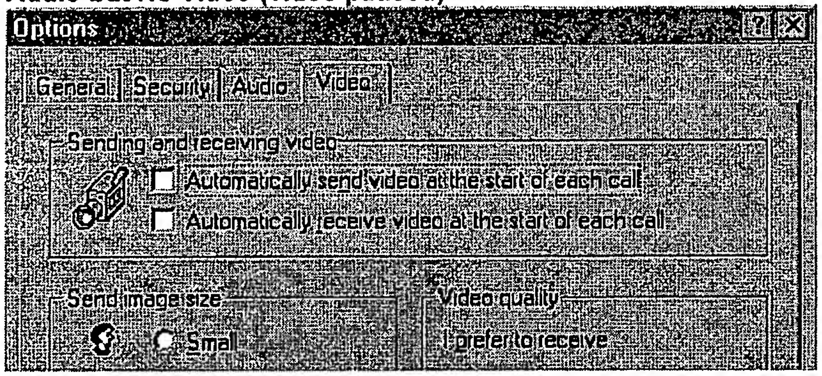
Hosting a Meeting not Required





Under normal circumstances the host meeting function is not required: All items in this dialog should be unchecked:

Audio but No Video (video paused)

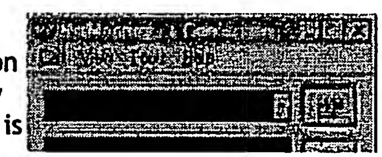


If your video setting in the ***Tools*** menu, ***Options*** item, ***Video*** tab is set to not automatically send or receive video when you start a call (a setting I recommend because it sometimes help establish the call faster), it will be necessary to manually start the incoming and outgoing video using the ***pause/play button*** at the lower left below the video window. Incoming and outgoing video can be paused or played independently by right clicking on the image (either the incoming image in the video window or your image in the picture-in-picture box or a detached "My Window") and checking or unchecking the ***Pause*** item.

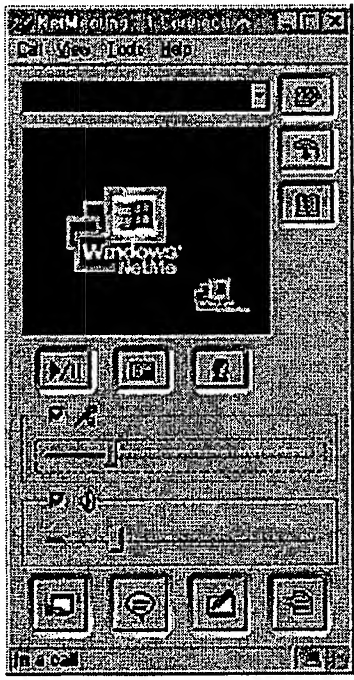
Incoming Audio No Incoming Video

If your settings are set to play incoming video and you are seeing nothing (but audio from the other party is heard) either the other party has no video camera or is not transmitting. You or they can pause the video incoming or outgoing by right clicking on the video and checking or unchecking the pause item.

You can determine whether the other party apparently has audio or video capability by making sure that you are in ***View Participant List*** mode. Clicking on an individual and selecting ***Properties*** will show you what audio/video capability the party apparently has. It is possible the even though audio or video capability is

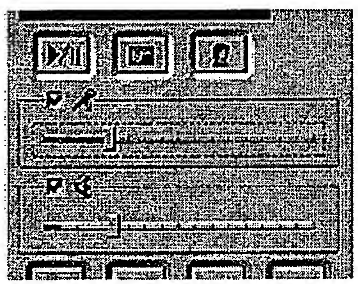


cannot effectively use it (i.e. they might have a TV card with no camera, a sound card but no mic)



If you instead have the audio meters view you can toggle between this view and the participant view using the button on the right below the video window.

Audio Problems



The audio meter view (available by clicking on the button on the bottom right of the video window if you are in the participant view) allows control of the audio properties of the call - the slider should be set at the lowest possible setting that will allow comfortable natural sound in both directions. The speaker section should have one green rectangle when the call is fully connected -- if it does not likely you are experiencing a problem with the H.323 connectivity caused by a router, firewall or proxy blockage (the Audio Tips page has a discussion) or the call was made as a data only call (see above).

If your microphone shows no meter movement when you speak likely either the microphone or its connection is faulty or your audio mixer is misconfigured (again the Audio Tips page has a discussion).

If the other party is experiencing echo of their voice and you are not using headphones (both you and the other end should be)-- it is possible to use your end in a half duplex mode by clicking off the microphone check box when you are speaking.

IEEE1394 Firewire cards and NetMeeting

IEEE1394(Firewire) cards (and DV cameras) do not usually provide the required drivers (with VFW interfaces) NetMeeting capture and thus cannot normally be used as video conferencing cameras.

At least one correspondent has found a (perhaps clumsy but usable) solution to the problem -- his comment is in response to my standard - it doesn't work response:

"Actually, this is not true. I have such a configuration and I have managed to use the camera with netmeeting, with help of a product named "SoftCam" from Luminosity Software (www.softcam.com), together with the "AMCAP.EXE" program, which is part of Microsoft's DirectShow SDK. Here's the setting: -amcap.exe runs in a corner of the screen and displays a real-time preview of the DV cam. It can be resized at will -When running Netmeeting, I choose the "SoftCam" video source. -In the SoftCam control panel, I select "live" mode, and make sure that the capture frame matches amcap's preview area.

As far as I can tell, it works as well as a "native" VfW device. The only thing is that you must remember not to clutter the capture area with another window. I agree that the setting is not very straightforward, and I would like to see a little piece of software that combines amcap and softcam into a single VfW driver that takes its source from any WDM video device. I don't see any technical problem in that, and I am surprised that it doesn't exist yet. Microsoft claims that it supplies a VfW-to-WDM mapper that is supposed to do exactly this, but it requires additional software from the hardware vendor."

Many individuals have used the Trackerpod software as a (clumsy but usable) work around -- it has a WDM (new style) to VFW (old style) capture interface translation.

Another correspondent indicates:

"It IS TRUE that DV cameras coming in over Firewire (or anything else) do not work because they typically do not come with the kind of drivers that Netmeeting uses.

It is NOT true that Firewire webcams do not work. I have four kinds of Firewire cards on which I can use video in all cases that I have tried. These are:

1) Audigy Firewire interface
2) PCMCIA Firewire (for laptops)
3) Orange Micro Firewire (three ports)
4) Orange Micro Ethernet/Firewire combo board

I have an Orange Micro iBot camera and have successfully used it with all of these boards on both Windows 2000 and Windows XP. I have not been able to use it on Windows NT because drivers to not exist for firewire that will handle a camera (drivers DO exist for IP). I have not used any of this on the Windows 98 platform because I do not have sufficient CPU power (it's an IBM ThinkPad 240 with a celeron) to run the video, but I suspect it would if I had a system with enough gas.

The advantages of the Firewire solution is that it can support a far higher data rate than USB 1.1. As USB 1.1 can do a

maximum of about 1mbps, it can not support a 30fps full color data rate. Thus, there is onboard circuitry on the camera does compression before sending the data over the wire to the host. The benefits of this are that the work of compressing the video data flow can be, in part, offloaded from the main system processor. That can be important with older systems.

The Firewire interface has a 400mbps data rate which will easily handle 30fps full color; no surprise as it was developed in part to handle DV camcorders. Since the host is getting the raw data, it can use more sophisticated methods to do compression which can also change as new drivers are introduced. Of course, you have to have sufficient CPU power to do this. I only run the Firewire iBot camera on either a 1.1 Ghz or a dual 900Mhz system.

I would recommend that you change the title to "DV Cameras and NetMeeting" as this would focus attention on the fact that is cameras without proper drivers. Firewire itself is simply the transport mechanism and doesn't have much to do with whether NetMeeting will actually work or not."

Problems, comments, questions: Email webmaster
Changed:Monday October 06, 2003 12:28 -0400

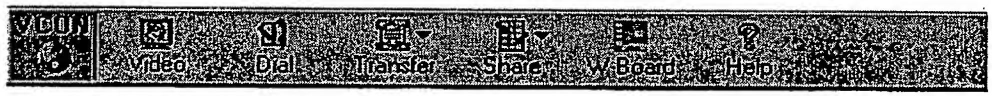
"Instructions on Application Sharing and Data Collaboration," VCON Escort and Cruiser,http://www.vide.gatech.edu/docs/share/

Instructions on Application Sharing and Data Collaboration

VCON Escort and Cruiser

Choosing the Data Collaboration Option

The VCON clients come with the capability of doing application sharing and data collaboration with either Microsoft's NetMeeting or with VCON's built in T.120 system. Therefore the first step in doing application sharing or data collaboration is to identify your choice. To do this, place your cursor on the Conference Panel (topmost gray bar.)



Press the right mouse and choose "Properties".

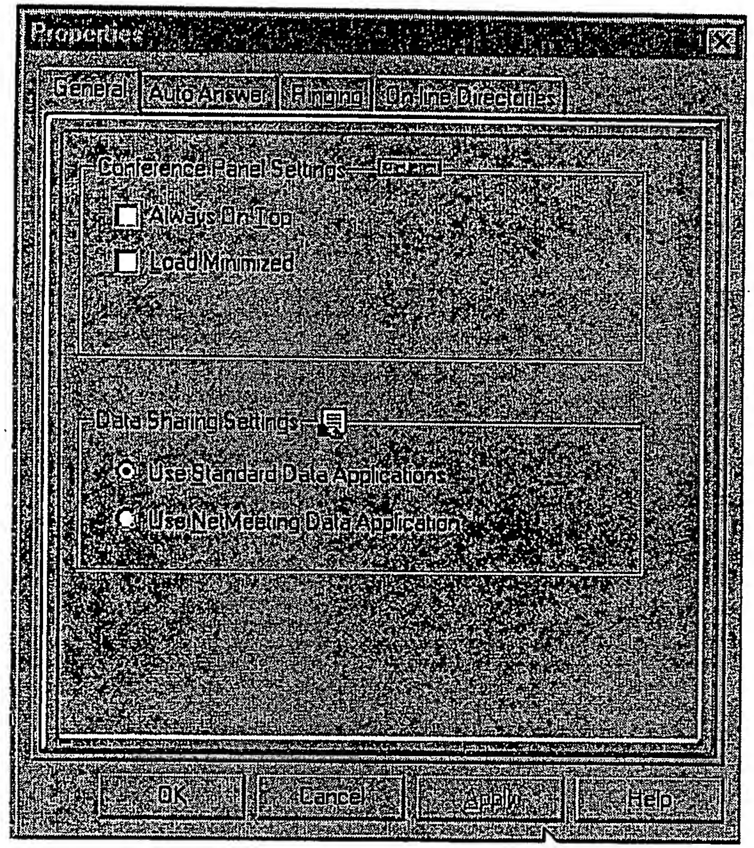
Here you will see several configuration options. You will find the "Data Sharing Settings" under the "General" tab. The VCON client gives you the option of using Microsoft's NetMeeting or it's own T.120 implementation which it calls the Standard Data Applications. Some testing and experience has shown several things:

- If the other client(s) in the call is(are) from another vendor, all clients should attempt to use NetMeeting.
- If only VCON clients are in the call, all should select the same Data Sharing Setting....either NetMeeting or the Standard Data Applications. Unpredictable cursor action/affect seems to result otherwise.

We will give examples of both in this document.

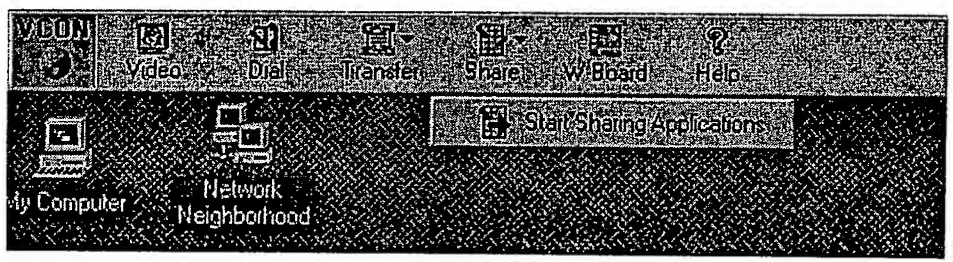
Data Sharing through the Standard Data Application

Click the "Use Standard Data Application" and then "OK". You will need to restart your client before the change actually takes effect.

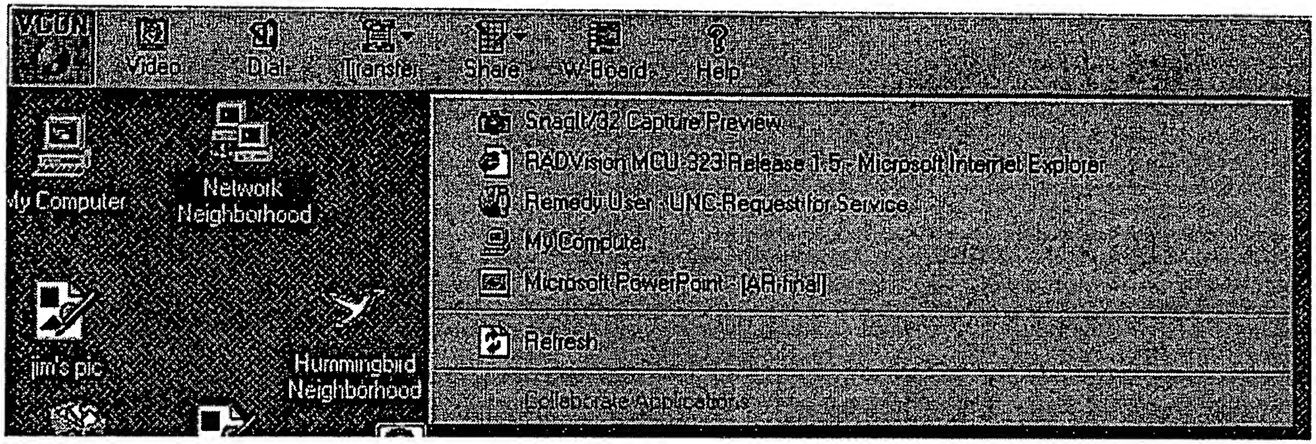


Establishing Application Sharing

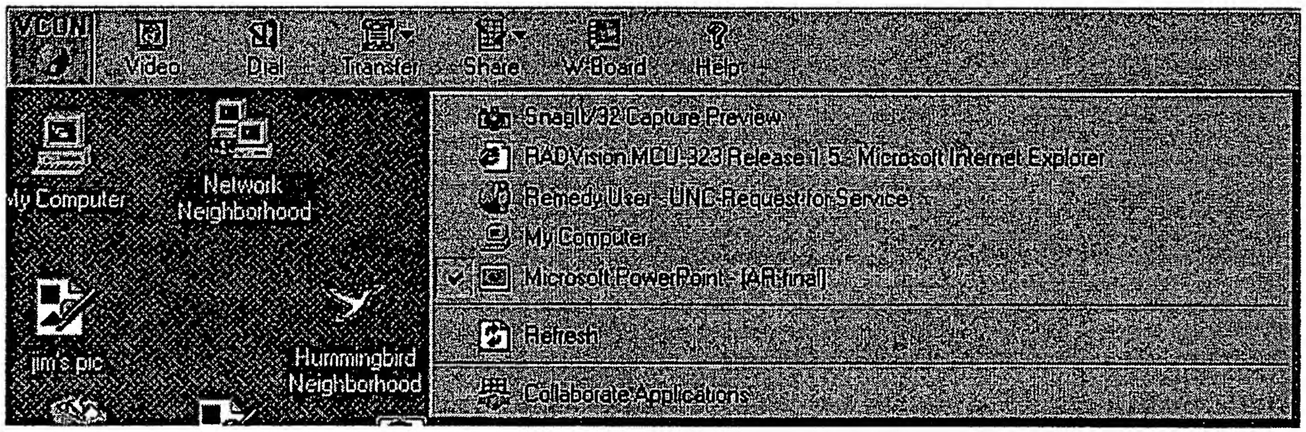
Either person can establish application sharing. We will call this person the "share-lead". The share-lead will go to the conference panel and click on "Share"



dragging down to highlight "Start Sharing Applications". Clicking on "Share" again will now show which applications are available for sharing.



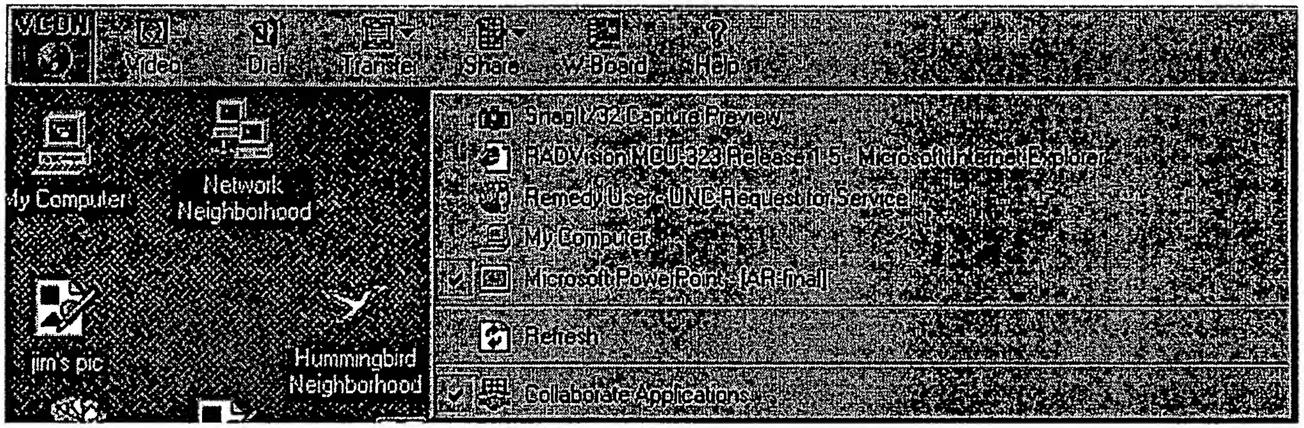
In this example, we will share a Power Point Presentation called "AR-final". To do his we drag down and highlight that entry. Clicking on "Share" again will now show



The Power Point application window and the open presentation will now appear on the other end station(s). The others will only be able to view the application and the share-lead's activities on it. (Note: If the share-lead puts any other, non-shared windows on top of the Power Point window, they will show on the other end station(s) as rather unsightly cross-hatched blocks of the same size as the unshared windows -- basically limiting or blocking their view of the Power Point.)

Establishing Data Collaboration

The share-lead can now allow the other end-station(s) to traverse through and modify the application by enabling the data collaboration mode. To do this, the share-lead will again click on "Share", this time dragging down and highlighting "Collaborate Applications".

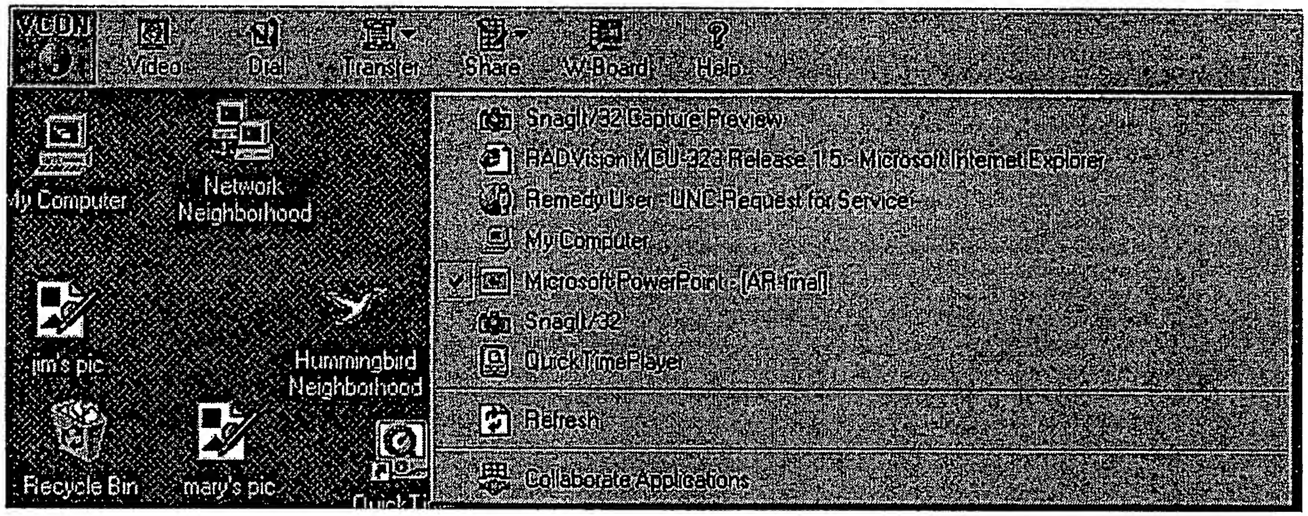


Now anyone in the conference can navigate through any shared documents, make changes to any shared documents, save or replace any shared documents, control the behavior of any shared applications. Several things should be mentioned:

- First, the application is actually running on the share-lead's machine. Any file activities will therefore happen on their machine as well. The other participants, for example, cannot save a shared document on their own machine. (See the "Transfer" button for moving files.)
- Secondly, for control of an application to transfer from one person to another, the new person can simply click their mouse anywhere in their screen. The cursor then falls under their control.
- And finally, while in data collaboration mode, there is only one cursor among all machines in the conference. It does not seem to matter whether someone is trying to use the shared application or a non-shared (private) application running on their machine. Should someone need to use the cursor (say to do a quick email check), proper protocol would seem to be "May I use the cursor for a minute" and, when finished, "Okay, someone else can use it." Unless such a protocol is followed, the collaboration could deteriorate into something we could term "Dueling Cursor" syndrome. We don't like it, we may report it. It doesn't seem to be a reasonable feature at this point - perhaps a configuration shortcoming and hopefully not a part of the standard.

Breaking the Collaboration or Sharing

To turn off data collaboration, simply click on "Share" and highlight it again. Notice that it is now unchecked.

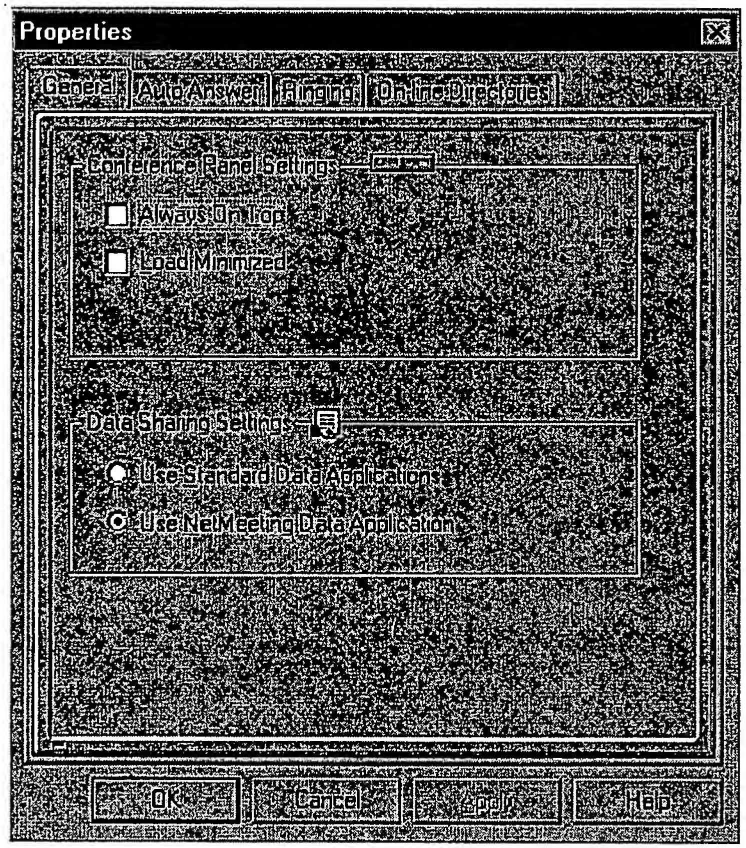


Application sharing is turned off similarly. Also notice that a new application has shown up on the list -- QuickTime Player. Anytime the share-lead starts up a new application, it will only appear under their "Share" button after they do a "Refresh" under this button.

At this point the participants can continue their conversation or hang up as usual.

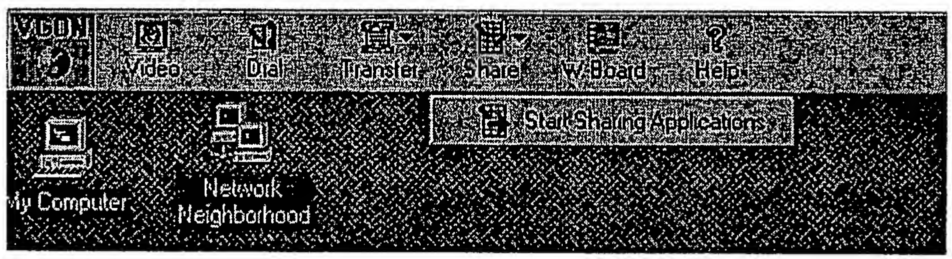
Data Sharing through NetMeeting

Click the "Use NetMeeting Data Application" and then "OK". You will need to restart your client before the change actually takes effect.



Establishing Application Sharing

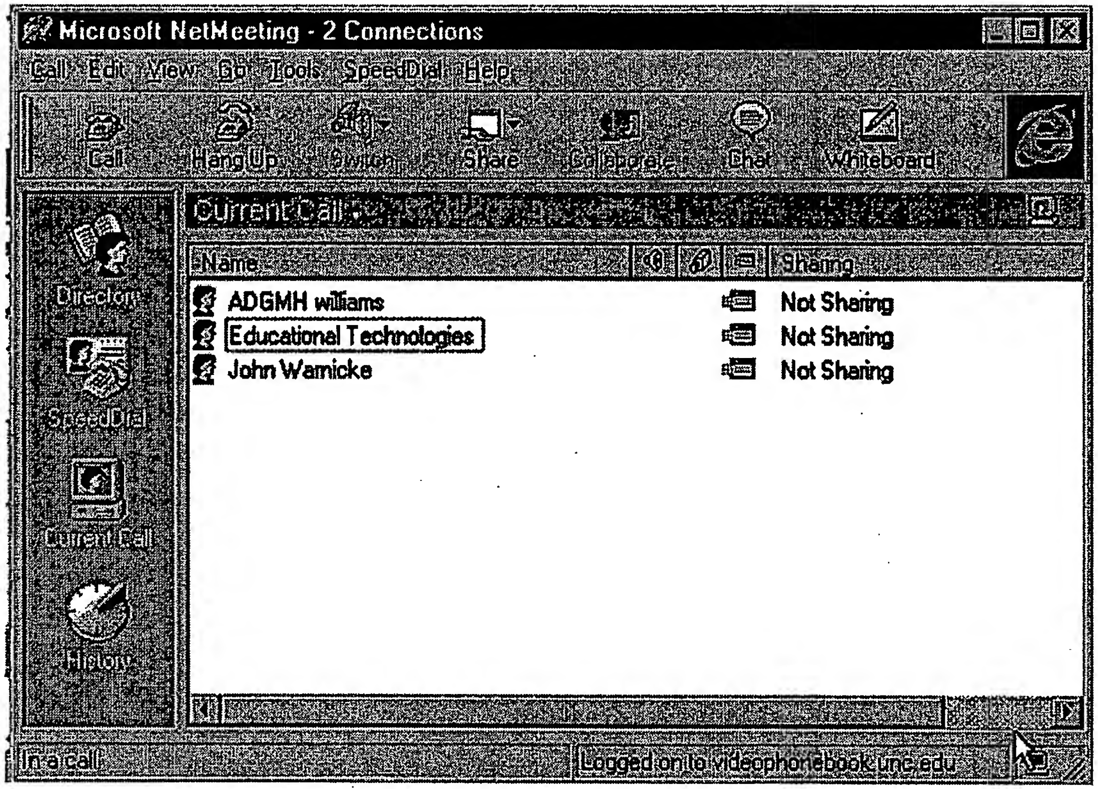
Either person can establish application sharing. We will, as above, call this person the "share-lead". The share-lead will go to the conference panel and click on "Share"



dragging down to highlight "Start Sharing Applications". This will cause NetMeeting to start on all workstations (meaning that it must be installed prior to the call.)

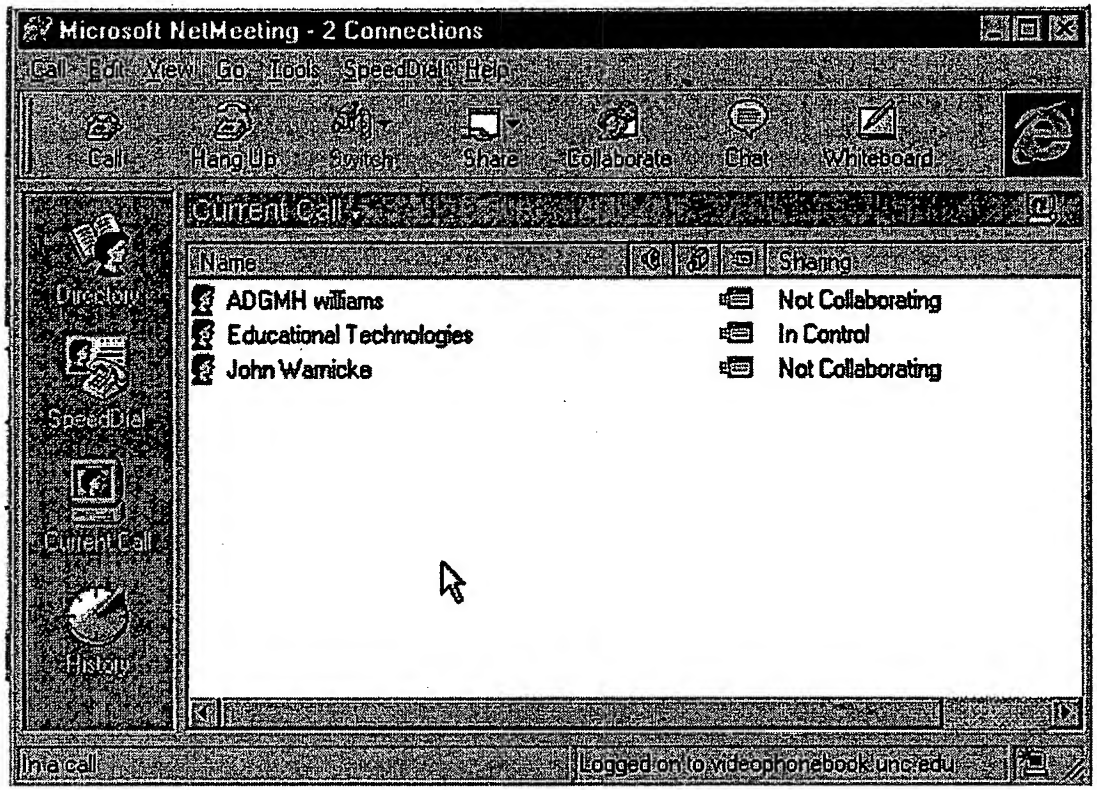
The NetMeeting Interface

At this point the video conferencing is happening through the VCON client and the application sharing and/or data collaboration is (or will soon be) occurring through NetMeeting. The NetMeeting window will initially appear as something like



At this point, no sharing or collaboration is taking place. The share-lead will start them by using the "Share" and "Collaborate" buttons on the NetMeeting window. Clicking on either will show which applications are available for sharing or collaborating. The share-lead will select any items they wish to share from this list.

Since the share-lead starts up the sharing and collaboration in this example, they are shown as "in control" initially (here it is Educational Technologies, ET for short.)



While in sharing mode, John and Williams will be able to see what ET has shared with them. If ET chooses to collaborate the application, John or Williams can take control of the application simply by clicking in the application window. For example, if John takes control of the application, the NetMeeting window would then change to show ET "Collaborating", John "In Control", and Williams "Not Collaborating". Williams can take control by clicking on the application window whenever his time comes to contribute information or operate the application.

As mentioned above for the Standard Data Applications, now anyone in the conference can navigate through any shared documents, make changes to any shared documents, save or replace any shared documents, control the behavior of any shared applications. Several things will be mentioned here as well:

- First, the application is actually running on the share-lead's machine. Any file activities will therefore happen on their machine as well. The other participants, for example, cannot save a shared document on their own machine. (See the "Transfer" button for moving files.)
- Secondly, for control of an application to transfer from one person to another, the new person can simply click their mouse anywhere in their screen. The cursor then falls under their control.
- And finally, while in data collaboration mode, there is only one cursor among all machines in the conference. It does not seem to matter whether someone is trying to use the shared application or a non-shared (private) application running on their machine. Should someone need to use the cursor (say to do a quick email check), proper protocol would seem to be "May I use the cursor for a minute" and, when finished, "Okay, someone else can use it." Unless such a protocol is followed, the collaboration could deteriorate into something we could term "Dueling Cursor" syndrome. We don't like it, we may report it. It doesn't seem to be a reasonable feature at this point - perhaps a configuration shortcoming and hopefully not a part of the standard.

Breaking the Collaboration or Sharing

Click on the Share or Collaborate button. Which items are actively being shared or collaborated are shown by the display of a checkmark to the left of those applications. The person who originated the sharing or collaboration can simply select any application (again) to toggle it off.

At this point the participants can continue their conversation or hang up as usual.

“Instructions on Multipoint Application Sharing and Data Collaboration,” VCON Escort and Cruiser with the RadVision MCU,http://www.vide.gatech.edu/docs/multi-share/

Instructions on Multipoint Application Sharing and Data Collaboration

VCON Escort and Cruiser with the RadVision MCU

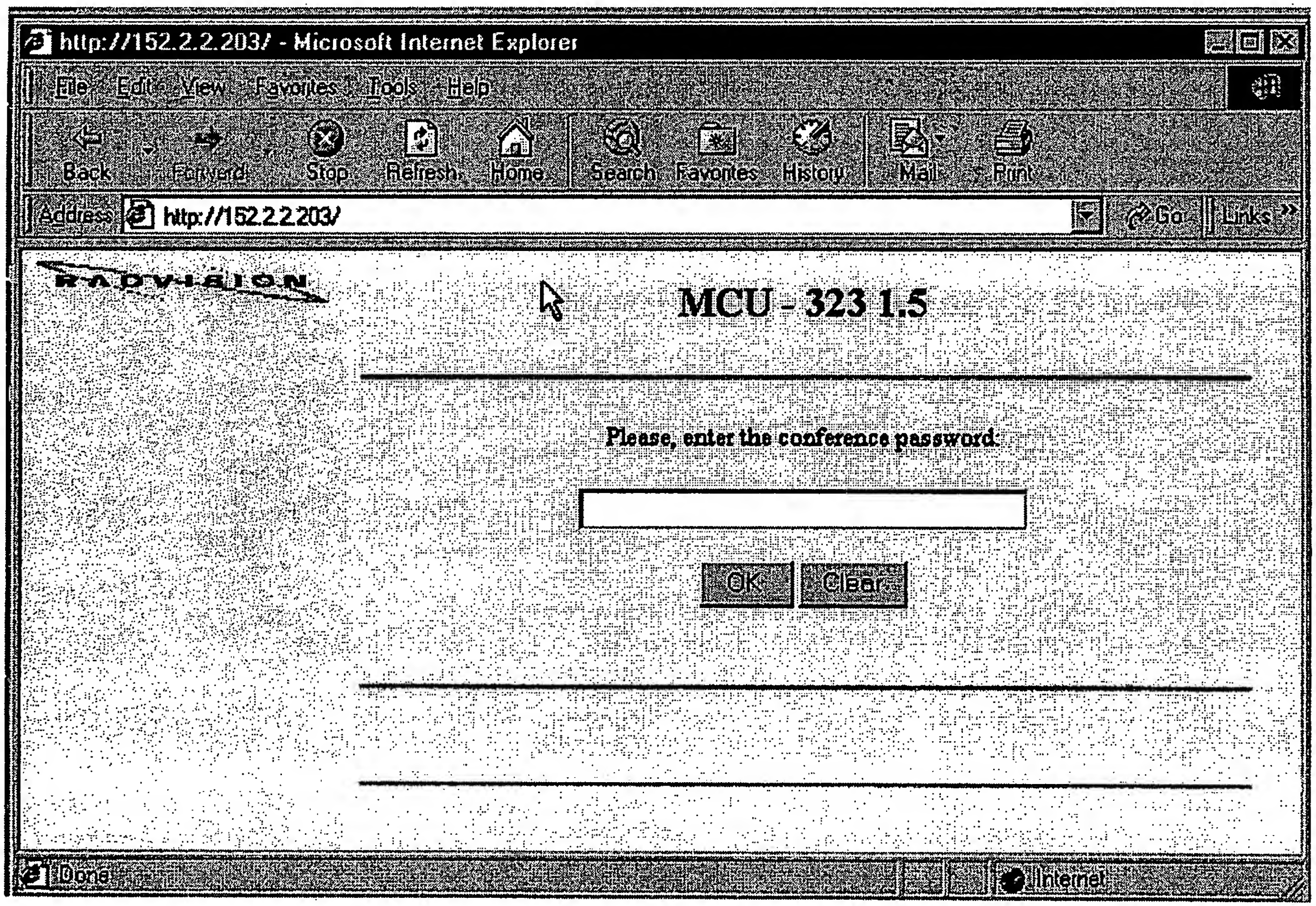
General application sharing and data collaboration instructions can be found in Instructions on Application Sharing and Data Collaboration. This material includes some additional information on what is required to start application sharing and data collaboration when in a multi-point call under a RadVision MCU and Gatekeeper.

Establishing the Data Share Group

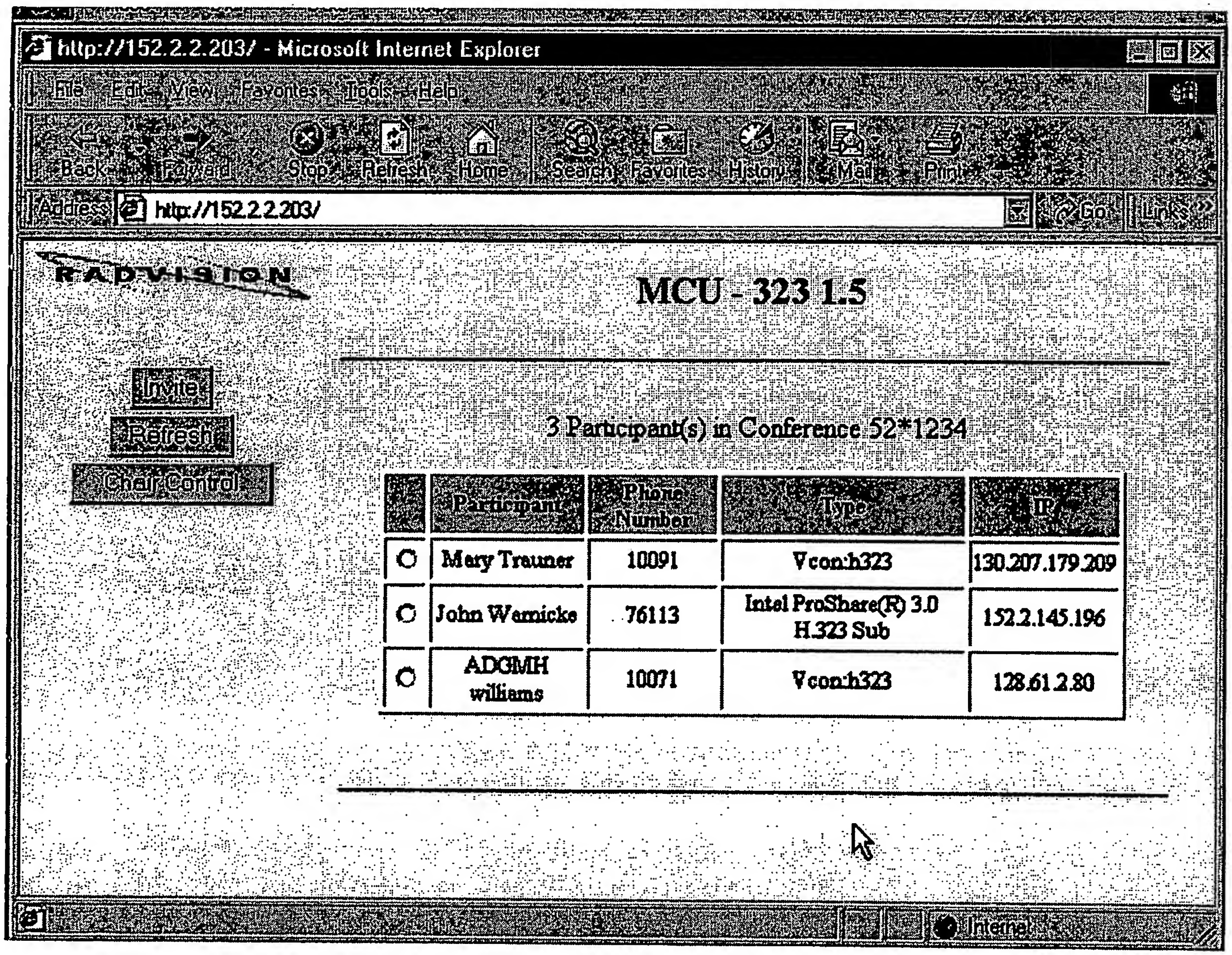
Since there will be three or more people in a multi-point conference, one person needs to bring all others into the MCU "Data Share" mode. (We'll call this person the share-lead.) To do this, the share-lead should start the Internet Explorer browser. (The interface does not work properly with Netscape; this *restriction* is being reported to RadVision.) The share-lead should then link to their MCU via its ip number. For this example

http://152.2.2.203

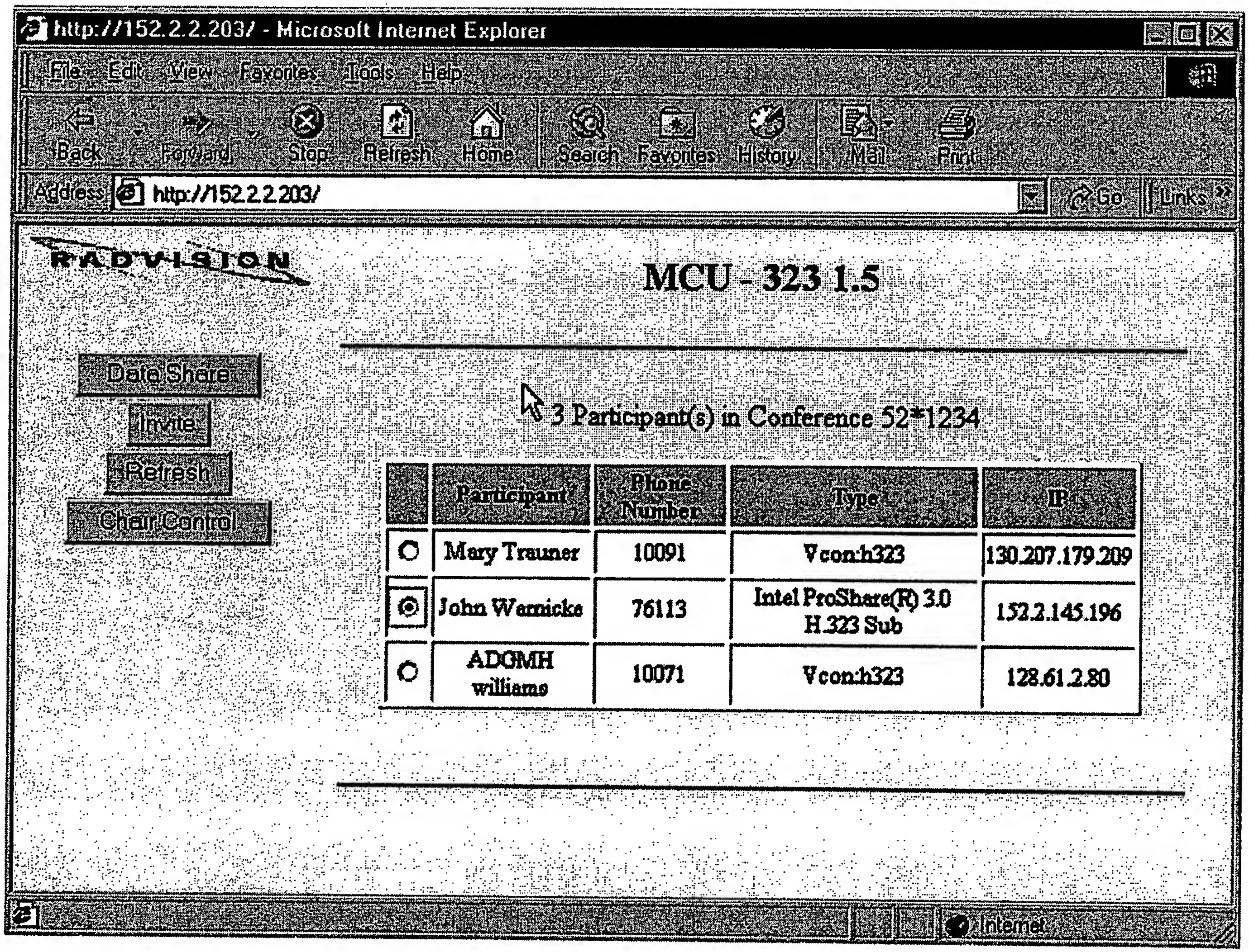
The MCU window will come up asking for a conference password.



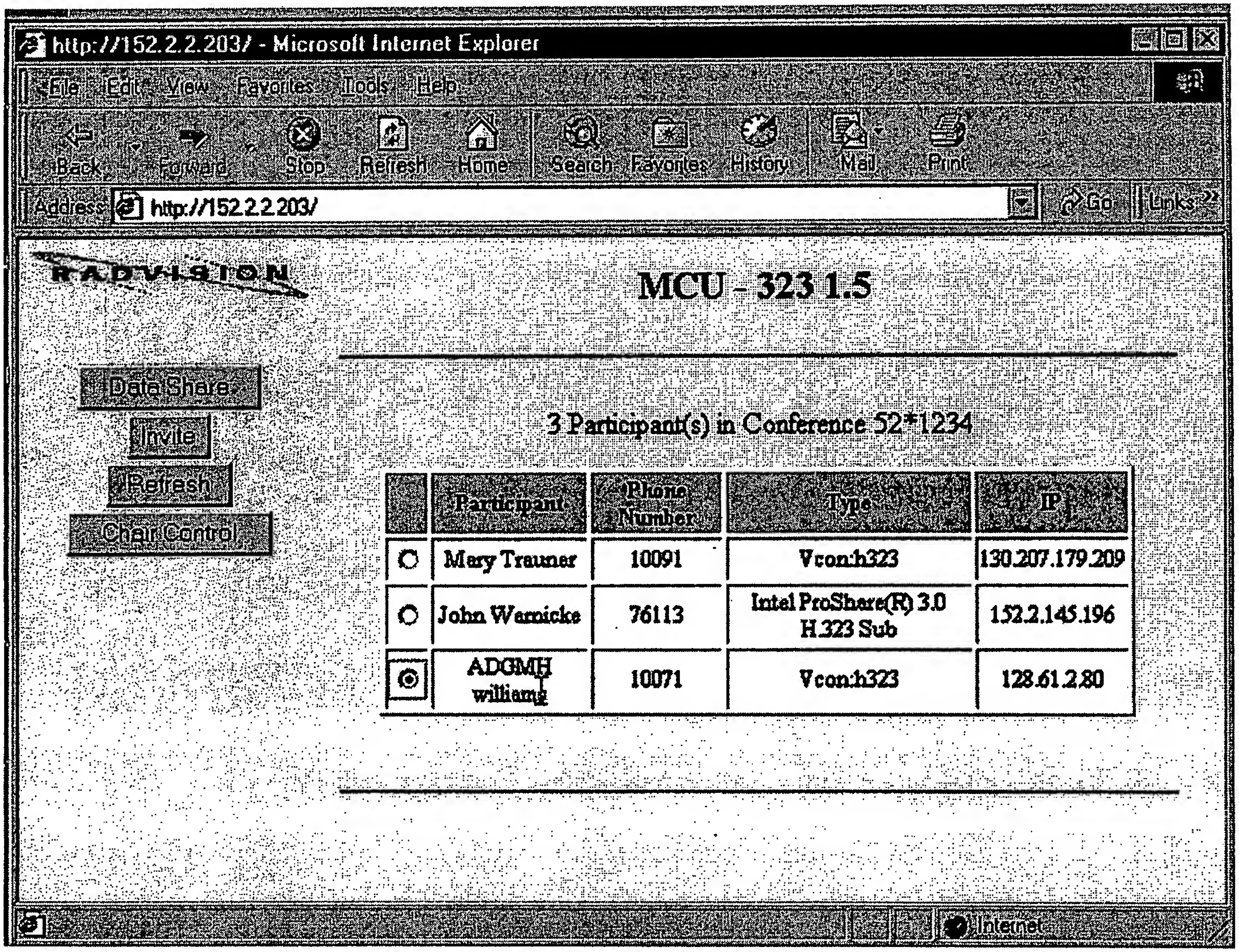
Enter that password (typically this is the same number that you dialed to reach the MCU) and click "OK". The window will show who is currently in the multipoint conference. (Note: This is not dynamic, therefore "Refresh" should be clicked every so often to see if anyone else has joined in or left.)



Each person currently in the conference is listed; a radio box appears to the left of each name. Let's say that Mary is the share-lead in this example. For each person with whom she wants to application share or data collaborate, she will click the radio box and then click "Data Share". First she will include John.



Then she will add in Williams.



If they are on VCON clients using the Standard Data Sharing, the VCON "Share" button scenario will be used. If NetMeeting is their application sharing choice, a NetMeeting window will appear on each screen as they are added to the Data Share. See Instructions on Application Sharing and Data Collaboration for further details on choosing applications and enabling sharing and/or collaboration under either of these options.

"File Transfer," Microsoft Windows Technologies Windows NetMeeting, last updated June 4, 1999,http://www.microsoft.com/windows/netmeeting/features/files/default.asp

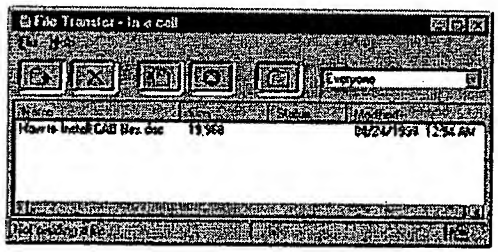
File Transfer

File transfer lets you send files during a NetMeeting conference.

With file transfer, you can:

- Send a file in the background to conference participants.
- Send the file to everyone in the conference, or to one or more selected participants.
- Accept or reject transferred files.

Learn more.



Last updated: Friday, June 04, 1999

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“From Dial Tone to Media Tone,” Analyst: R. Mahowald, IDC, June 2002

From Dial Tone to MediaTone

How the WebEx Interactive Network Powers Business Communications to New Heights

Analyst: Robert Mahowald

OVERVIEW

While audio conferencing and static communications (e.g., email and telephone) rivaled face-to-face meetings as the most important forums for business meetings in the 1990s, Web conferencing — with its real-time multimedia communications, data sharing, and computer-telephony integration (CTI) — is poised to drive business communications in the new millennium. The dramatic uptake in demand means new opportunities for vendors of Web conferencing products.

IDC research points to the rapid adoption of Web conferencing in areas as disparate as sales, marketing, training, support, engineering, channel management, and internal employee communications. Findings from IDC's *2001 Conferencing Survey* show that current and planned buying is strong: An average of 43.6% of respondents reported they plan to increase their conferencing usage by 100% or more in the next six months, whereas only 1.5% plan to do less conferencing in the next six months.

But what is pushing the user numbers ever higher is the fact that Web conferencing is fast becoming a general-purpose communications service. Business users are increasingly looking at online conferencing to more effectively communicate with customers, prospects, suppliers, and partners. They are also using the technology in a way they had formerly reserved for the office water cooler — as a proxy for a gathering "place," a notional room for spontaneous conversation, discussion, and planning. People who entered the workplace during the 90s are as used to the PC interface as they are familiar with the telephone, and they display much the same comfort and facility with conducting a visual conversation using a computer as they do a voice conversation with a telephone handset.

While current use of Web conferencing is robust, IDC forecasts an even stronger future for this market. This growth prediction is based in part on the assumption that applications and services will be deployed on networks that are increasingly reliable, scalable,

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extensible, global, and cost efficient. As Web conferencing enters prime time in the enterprise, and vendors seek to differentiate themselves with new features, customers will not overlook the importance of the CTI networks that provide the open systems interconnect (OSI) layers critical to highly available, rock-solid conferencing systems.

IDC believes that MediaTone — and the WebEx Interactive Network it powers — is an example of the kind of "guarantee" that conferencing vendors and service providers need to make to customers.

WebEx's new multimedia switching platform is driven by the company's MediaTone communication signaling technology for sharing multimedia information and datastreams. IDC believes that MediaTone — and the WebEx Interactive Network (WIN) it powers — is an example of the kind of "guarantee" that conferencing vendors and service providers need to make to customers. Among all the other differentiators, the ability to provide "dial tone"-like quality of service assurances is part of what will take Web conferencing to new heights.

METHODOLOGY

This white paper highlights the opportunities for vendors of conferencing and related collaborative multimedia services. Further, this paper presents one vendor, WebEx, and its deployment platform, WebEx Interactive Network, as an example of a solution that addresses this opportunity.

This paper's focus is qualitative rather than quantitative; that is, we do not attempt to quantify the size of this market opportunity. Instead, we discuss the tremendous opportunity for vendors in this market segment and how clients can best leverage this opportunity.

INTRODUCTION

How are businesses meeting, communicating, and sharing information? The answer is complex. The kinds of meetings, the contexts, the participants, the information shared, and the results are evolving as technology races to develop ways for visual collaboration to rival — and sometimes be more efficient than — face-to-face meetings.

If we think about how verbal vocabulary has evolved to meet the changing needs and mores of its users, it is clear that it could only do so because it had a flexible but unchanging semantic structure — subjects, verbs, sentences, clauses — on which to hang the words.

If we think about how verbal vocabulary has evolved to meet the changing needs and mores of its users, it is clear that it could only do so because it had a flexible but unchanging semantic structure — subjects, verbs, sentences, clauses — on which to hang the words. As the lives of early humans grew more complex, more words were added to the language to accommodate the dynamism of the communication. This growth has progressed to the present day, and we now have more words than we can ever use to describe a myriad of complex situations, events, and concepts.

Communications technology has evolved along the same lines. The telephone, for example, has a basic structure. Our plain old telephone system (POTS) — physical cables, network switching centers, and

end-point devices in homes and offices — has provided a basic structure on which to build not only simple telephony features such as the ability to call any number in the world but also complex management features such as call waiting, voicemail, unified messaging, and conference calls. Like language with its structure, shared understanding of words, and ubiquity, the POTS had to be utterly reliable, simple to use, global in reach, and in almost every home and office.

It is important to understand the real vision of the builders of the original POTS voice network. They realized that they needed to build something tremendously flexible, extensible, and standards-based to accommodate any kind of innovation in the years to come.

It is important to understand the real vision of the builders of the original POTS voice network: Even though these early engineers had no idea what kind of voice services would be introduced in later years, they realized that they needed to build something tremendously flexible, extensible, and standards based to accommodate any kind of innovation in the years to come.

Although in 1880 there may have been some doubt as to whether Marconi's telegraph or Bell's telephone would win the right to play a seminal role in the lives of nearly every human being, the network built around the telephone, and the network effect to produce millions of network nodes (i.e., phones), brings us to where we are today.

We live in the age of the Internet. Corporate workers have become used to interacting with their PCs almost as they would with a colleague; much as the telephone handset became a physical proxy for the person on the other end of the line, PCs are an embodiment of how we in business communicate today, with email, instant messaging, calendaring and scheduling, and team collaborative applications. Browsers are increasingly our information communication devices, more than the telephone.

When CTI became possible in the early 1990s, technologists looked for ways to make a tighter link between data and voice and to build CTI products as reliable, useful, and ubiquitous as the telephone network. To date, CTI has brought us the uniting of voice, fax, and email messages in a common object store (unified messaging); interactive voice response (IVR); and numerous other linkages between the PC and the telephone.

But these are largely one-way products. They rely on the PC as a viewing device, or as a window into a network object store, and little more. Email is store-and-forward technology, unified messaging is "push" technology, and even instant messaging, with all its connotation of speed, cannot match the free-flowing spontaneity of a lively conversation.

MORE THAN MEETINGS

Both the profile of the typical Web conferencing user and the business areas in which Web conferencing is being used have evolved dramatically in the past few years. As conferencing services vendors deploy more specialized, high-touch features to address more meeting types, their use as substitutes for existing types of online communications has grown. Just three years ago, most conferences were audio only, and data sharing was a one-to-many activity, with limited user controls and no flexibility.

Today, the average user is not an "IT type." Users of most Web conferencing products don't have to go to special conference rooms or schedule time on the system with a gatekeeper in their company's telephony department.

Today, the average user is not an "IT type." Users of most Web conferencing products don't have to go to special conference rooms or schedule time on the system with a gatekeeper in their company's telephony department. As the technology has grown more linked to the desktop PC, control over meetings has been decentralized. Scheduling, attendance, richness of the information shared, and other issues are decided largely by the meeting's host, and users don't need to schedule a meeting so much as click a button to launch an ad hoc session.

How have meetings changed? Only a few years ago, many meetings were audio-only conference calls, handled by incumbent local exchange carriers (ILECs), and they were scheduled using a flood of telephone calls. When users needed to see something, they had to congregate in a single physical location. There was a place to go to for meeting and another place to go to for working. Users did their thinking offline, asynchronously, then arranged a call to convey decisions already made and strategy already formed.

Today, electronic meetings are about collaborating, sharing, and teaching. They are part of the process, not the end of the process.

Today, electronic meetings are about communicating, sharing, and teaching. They are part of the process, not the end of the process. Think of the new uses for conferencing today: salespeople conducting meetings with prospects and existing customers, customers and employees receiving training anywhere in the world, and experts providing live hands-on support to remote customers. While the first phase of business communications was face to face, the advent of the telephone enabled most communications to be remote, with fewer face-to-face meetings. This change dramatically increased business opportunities. Now, with the ubiquity of multimedia-based Web communications, users can rely on both telephone and visual capabilities to dramatically enhance the effectiveness of remote communications. This has the effect of further reducing the cost of doing business and more significantly increasing business opportunities, just as the telephone did over the last century.

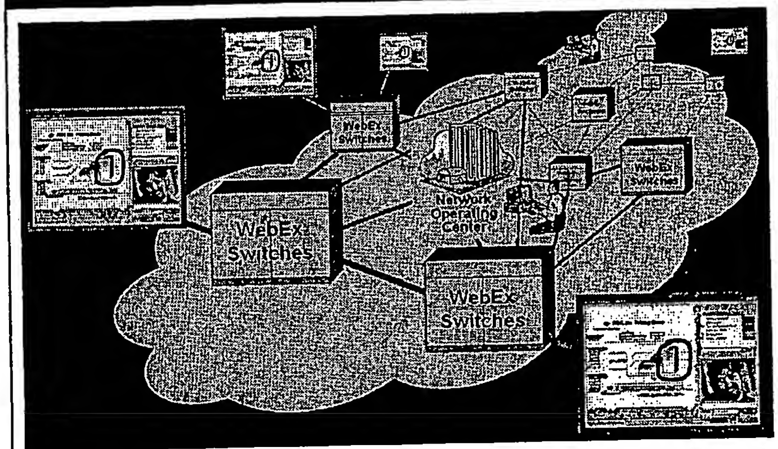
Salespeople use Web conferencing to present and deliver online demonstrations to customer prospects across the country. Instructors can reach students anywhere there is a Internet connection, with guided learning, feedback, and performance assessment — just like in the classroom. Engineers can share ideas with 3D computer-aided design (CAD) objects and get quick feedback. At one end, marketing can present new products to thousands of prospects, and at the other end, two colleagues can share the beginnings of a great new idea born from digital scribbles on a white board. Welcome to the wonderful world of Web conferencing.

WHERE NETWORK ARCHITECTURE FITS IN

Architecturally, conferencing links voice and data by providing a switching network that unites the public switched telephone network (PSTN) and the public Internet. The standard OSI model describes the seven layers on which communications data must be addressed for it to be successfully transported — from the miles of cables buried under highways and skyscrapers, through data and network synchro-

nization, session initiation, presentation (encryption and conversion), and, finally, up to the application layer, the user interface, and the PC (see Figure 1). It is a complex model, and while it is possible to combine certain layers, a weak link, or skipped layer, could mean a dropped call, a misrouted data file — in short, failed communication.

Figure 1: Network Architecture



Source: WebEx, 2002

One can imagine the machinations that must go on behind the scenes to produce a successful online meeting: As the audio portion moves along the PSTN, it passes several possible points of failure. Instructions from the user interface need to be transmitted via the packet-switching OSI layers — application, presentation, and session — where audio encounters Secure Sockets Layer (SSL) data encryption, decryption, and data conversion. Then, these two streams — audio and visual — have to meet seamlessly via a PSTN bridge and be transmitted to many users — all in real time.

When the telephone first emerged, Bell Systems dealt with this complexity by focusing on the redundancy and failover capacity of its network. The result is that the dial tone has become the sonic metaphor for global reach, utter reliability, high quality, and security. The network for Web conferencing has the same business requirements and attributes as the telephone system: Users want to be able to plug in a PC or mobile device and go, with the assurance that they will have dial-tone reliability no matter where on the Web their meeting takes them.

An added point of complexity comes from the fact that while the PC industry is relatively nascent, true CTI is even more new. Even top technologists don't know the bounds of the Internet, and the borders between data and voice are made ever fuzzier as voice moves to the Internet with voice over IP (VoIP). Because the horizon is always shifting, the network on which communications services are built needs to be immensely extensible — blind to the vagaries of different operating systems, platforms, devices, user types, and so on. It needs to be generic enough to not get in the way of the changing

uses, mobile platforms, and shifts in technology that are sure to come about in the next 25 years.

WEBEX OVERVIEW

WebEx (NASDAQ: WEBX), based in San Jose, California, provides a communications infrastructure for real-time business meetings on the Web. WebEx's products are carrier-class communications services that integrate voice, video, and data to enable collaboration, information and process sharing, and training. These services are based on WebEx's multimedia switching platform. WebEx services enable end users to share presentations, documents, applications, voice, and video on Windows, Macintosh, or Solaris systems, and they can be accessed via a Web browser. These services are used across the enterprise in such functions as sales, support, training, marketing, and engineering.

WebEx is now delivering these services to more than 6,100 corporate accounts through 200 different distribution channels.

NETWORK TOPOLOGY AND THE WEBEX INTERACTIVE NETWORK

When you understand how visual meetings, conferences, and etraining sessions are delivered, the dial-tone analogy really begins to ring true. Web meetings are a Web-based extension of audio communications, and in almost all cases, audio accompanies the visual portion of the session. Audio can be delivered via the network of a local telephone service provider or a larger carrier such as AT&T, France Telecom, or WorldCom. In many cases, voice services are delivered by the provider of the visual conferencing, or they can be offered using Internet protocols (IP), so that the POTS is bypassed entirely. Many current Web conferencing industry leaders allow for a variety of options, and participants on a call may use a combination of POTS and IP voice to synch up with the visual portion of the meeting.

The key point is that providers are gradually moving more of the controls for these services to the Web interface. Scheduling is increasingly accomplished via a browser-based interface. Dual-tone multifrequency (DTMF) controls within the Web interface let users manage and manipulate the audio part of the call using Web consoles. Because Web conferencing began as a marriage of new companies offering Web-based software products and services and existing vendors selling telephony services, it is tempting to think of Web conferencing as strictly a software business, with the telephony integration an afterthought.

Just as the original POTS architects needed to build a foundation for any future innovations in voice services, engineers building today's data networks find themselves faced with complex choices about building a network to deliver combined voice and visual services.

But telephony integration is perhaps the most intricate piece in the whole conferencing puzzle. Just as the original POTS architects needed to build a foundation for any future innovations in voice services, engineers building today's data networks find themselves faced with complex choices about building a network to deliver combined voice and visual services.

To deliver these communications services reliably and with predictable global performance, WebEx has deployed its own global network: the WebEx Interactive Network (WIN). WIN is composed of the application, presentation, and session functions (OSI Layers 5–7), with high-speed connections to the Internet on one end and voice bridges to the PSTN on the other end.

WIN is a network of WebEx multimedia communication switches that are architecturally distributed and highly scalable. The WebEx switches have substantial innovations that are rooted in WebEx's philosophy that conferencing is essentially a communications service, not a software application, and therefore has the same requisite business requirements and quality of service demands as the telephone network.

By leasing IP lines worldwide, WebEx has created a fully meshed network to ensure fault tolerance and rerouting in times of high use. The company has leased lines connecting communication hubs at colocation facilities that are distributed across the United States, Europe, and Asia/Pacific and has been expanding its global reach. Configured as a distributed network similar to phone networks, each hub is architected to be scalable and extensible. Each communication hub contains clusters of switches, ensuring high levels of reliability, redundancy, and scalability. As hubs are added,the regional hubs will act as contingency sites for each other, delivering what WebEx terms global "rings of service." WebEx's network will provide continuous reliability benefits because each additional node provides additional capacity, added paths for reliability, and reduced reliance on public networks.

Through WIN and the WebEx switching platform, information can be shared globally in an instant.

Through WIN and the WebEx switching platform, information can be shared globally in an instant. This guarantees reliability by controlling the network and by automatically routing and rerouting information based on network performance. Participants connecting with WebEx are automatically connected through the nearest WIN communications node, eliminating numerous network Internet service provider hops that are typically required. Should performance through a particular communications hub begin to degrade, alternate regional servers will be automatically pinged, and the next request will be routed through the nearest alternate server with the best performance level. These routing and rerouting capabilities ensure high levels of service for WebEx customers. These measures allow WebEx to guarantee fast and accurate delivery of information, strengthening the company's ability to alter current business methods.

Although WIN has been continually optimized and upgraded since its introduction two years ago, WebEx is moving toward even greater guarantees of reliability and scalability with the announcement of its major communications services infrastructure upgrade, MediaTone.

MediaTone

The Internet is increasingly crowded and unreliable. Packets of data and voice information may leave a gateway with great speed and determination, only to be blocked by least-common routing and Internet traffic. Packets may arrive at their destinations scrambled, late, or not at all, and because of the added complexity of bridging voice from a separate (PSTN) system onto Internet data, the points of possible failure become almost infinite.

Building on its existing support of a rich set of data, video, and audio capabilities, MediaTone has enabled WebEx to provide new capabilities that only a switched based network infrastructure can deliver.

WebEx's new MediaTone switching technology allows the WebEx switching platform to share complex media types, deliver advanced communications functionality, and support a range of new devices and platforms. The MediaTone signaling technology is part of Layer 6 (presentation layer) of the IP-based communications network layers and specifically provides the capability for real-time delivery and synchronization of multimedia content. Building on its existing support of a rich set of data, video, and audio capabilities, MediaTone has enabled WebEx to provide new capabilities that only a switched based network infrastructure can deliver, including:

- Support for Universal Communications Format (UCF) (UCF is WebEx's protocol for sharing rich media content within PowerPoint presentations in a way that lets users completely control the delivery. Full animation support is provided.)
- Sharing of embedded Flash files, with the ability to control start, stop, and pause files
- Sharing of Windows Media Player/RealPlayer content, with the ability to start, stop, and pause delivery to all participants
- Sharing of CAD and other 3D objects, with full manipulation
- Sharing of previously recorded WebEx meetings
- Secure access to or sharing of information in a meeting, whether the content is local or remote
- Multipoint videoconferencing, from either a browser or in support of Polycom cameras and standard video camcorders
- Access to presentations using handheld and wireless devices, with the ability to participate in meetings
- Simultaneous sharing of multiple documents or presentations; viewing of multiple documents at the same time, with the ability to flip back and forth between them

But is all of this really important if, as we said earlier, the average user is not an IT type? Web conferencing technology has trickled down from IT to desktop users, and it is now a part of the lives of most knowledge workers. Users see a user interface, and they have user-operated controls. Why is the network important to them?

All the bits and pieces assembled to link the PSTN and the Internet and to deliver voice, video, and data reliably, globally, and flexibly are important because if a system is down just once, drops calls, or doesn't accommodate specific situations and needs, then it is dead in the water. Disenchanted customers are like flowing water: They will find another path to their goal should one way be blocked.

WEBEX COMMUNICATIONS SERVICES

WebEx offers multiple communications services to fulfill diverse user needs within enterprises. Services include:

- **Meeting Center** provides rich interactive meeting environment.
- **Event Center** (formerly OnStage) enables delivery of multimedia Web seminars.
- **Support Center** (formerly OnCall) enables delivery of remote, hands-on technical support.
- **Training Center,** which is WebEx's newest service and the first to be based on MediaTone, enables remote delivery of live training to customers and employees.

CHALLENGES

A key challenge to overall adoption of Web conferencing is that several related applications and services deliver parts of the promise of real-time conferencing without all the pieces. For example, for most knowledge workers, the inbox paradigm is very strong. For small businesses, or firms with employees, partners, and customers relatively proximate, email, telephone, and other means of communication may continue to suffice. For larger businesses, where customers, prospects, partners, and employees are more geographically dispersed, however, communications beyond the telephone are essential.

The true barriers to entry in the market are far higher for vendors seeking to serve the marketplace as global communications service providers.

At first glance, the cost of developing Web conferencing products seems low. A few employees writing code can write a Web application server to store and share limited forms of information. The true barriers to entry in the market are indeed far higher — especially for vendors seeking to serve the marketplace as global communications service providers. Worldwide reach and reliability require a global network with peering agreements, impeccable code, built-in security, high scalability, rich functionality, and an army of client service representatives to meet the standard for 24 x 7 service that is increasingly expected of conferencing service providers. WebEx has had to invest substantially in its switching platform, WIN, and MediaTone signaling technology in the past few years, and the cost of acquiring new customers remains high.

Vendors such as WebEx face competition from a growing pool of Web conferencing product and services suppliers.

WebEx's continued demonstration of dial-tone reliability is important if it is to convince new customers that it can deliver the kind of Web conferencing capability and uptime they require.

CONCLUSION

The promise of dynamic, cost-effective, real-time, and productive Web-based communications is real. IDC's *2001 Conferencing Survey* revealed that 82% of businesses with more than 500 employees use some type of conferencing application. Buyers will continue to be drawn by the tremendous potential return on investment that they can realize through savings on equipment, bandwidth, personnel, and applications licenses, as well as sharply reduced travel costs.

But the real promise of Web conferencing is that it takes the dream of the telephone and extends it far beyond voice. Customers expect rapid responses. Competition pushes businesses to plan, design, implement, and support products on a 24 x 7 basis. Therefore, talk is not enough. To lead, businesses need to show, link, exchange, demonstrate, teach, relate, and collaborate. As business cruises along at Internet speed, users will increasingly need to share rich digital files, demonstrate products, confer with colleagues, and teach customers in real time. And they will need to perform such tasks from multiple types of emerging portable IP devices.

WebEx provides a comprehensive solution based on its communications services, infrastructure, and worldwide network. Demand for these services will continue, and reliable, scalable, extensible, and ubiquitous services such as WebEx's will be among the winners at the forefront of this shift as adoption of real-time multimedia communications continues to grow.

The compelling nature of what WebEx has accomplished is reflected in its having earned 5,000 corporate customers and more than doubled revenue in each of the past two years.

The compelling nature of what WebEx has accomplished is reflected in its having earned 5,000 corporate customers and more than doubled revenue in each of the past two years. WebEx's reseller relationships with a number of key telecommunications vendors (including AT&T, MCI WorldCom, NTT, France Telecom, and Telia) highlight the reputation WebEx commands. WebEx's approach — looking at Web-based, real-time, interactive Web services as a communication technology and infrastructure issue rather than a software tool or application — will enable both the company and its MediaTone technology to power business communications to new heights.

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“MediaTone - The ‘Dial Tone’ for Web Communications Services,”
Webex, 2003

webex™

MEDIATONE™

The "Dial Tone" for Web Communications Services

The "Dial Tone" for Web Communications Services

We've got to start meeting like this.

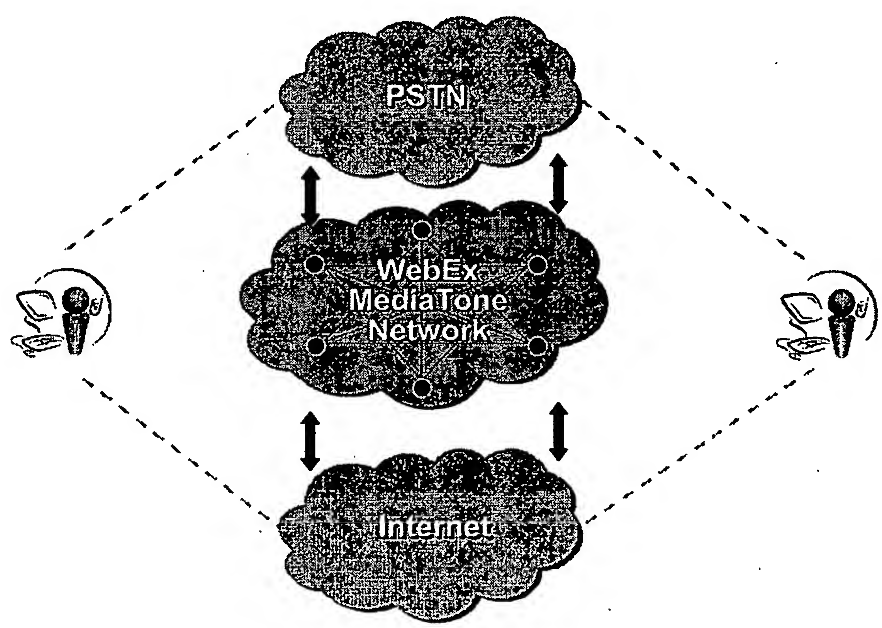
WebEx™ MediaTone™ technology enables WebEx meeting participants worldwide to enjoy the richest set of data, voice and video interactive services together with unparalleled network performance and reliability. The secure, highly scalable MediaTone Network can support millions of simultaneous calls, and as many as 5,000 individuals may attend one meeting.

The WebEx MediaTone Platform positions WebEx as the technological leader in Web communications services, providing OEM services to WebEx partners and industry-leading meeting services to nearly 7000 corporations. The MediaTone architecture provides ubiquitous access—regardless of location, hardware platform, operating system, browser, and wired or wireless status—enabling everyone to reap the benefits of online meetings.

MediaTone Technology

All WebEx services integrate the company's MediaTone technology. This proprietary technology enables true interactive communication sessions with levels of functionality, reliability, security and scalability impossible to achieve in a database-centric, store-and-retrieve architecture. With its modular framework and standards-based application programming interfaces (APIs), MediaTone is the "dial-tone" for Web communications.

The highly extensible WebEx architecture includes two components: the MediaTone Network and the MediaTone Platform.



Customer Requirements

As they evaluate solutions that can deliver such benefits, customers factor in many technical and business requirements. These include the need for:

- A flexible, scalable, reliable Web services architecture with powerful and extensible capabilities.
- The ability to accommodate millions of people-minutes in online meetings.
- Reliable third-party service provision for integrated rich-media calls across enterprises.
- Integration with the customer's enterprise IT infrastructure and support for heterogeneous environments *(eg. Microsoft Exchange/Outlook and Domino/Notes integration, Instant Messaging, Portals).*
- Support for diverse online meeting scenarios (e.g., sales, marketing, training, support, project management, design reviews).
- Rapid deployment and ease of administration.
- Security that addresses authentication, encryption, auditing and tracking.
- Full support for ad-hoc and scheduled sessions.
- Integration with billing and reporting systems.
- Support for personalization.
- Seamless integration with voice systems such as audio conferencing bridges and Voice over IP (VoIP) solutions.
- Worldwide 24x7x365 support and native language support.

MediaTone Network

The MediaTone Network is a fully redundant, high-performance private global network specifically designed to deliver Web communications services. Created with a carrier-class information-switching architecture, the MediaTone Network delivers optimal performance by routing communications across several WebEx data centers. The result is a high-performance network that is unmatched for secure, reliable, fast, real-time Web communications. WebEx is the only company to develop and deploy a globally distributed information-switching network specifically designed for the delivery of interactive Web communications services.

MediaTone Platform

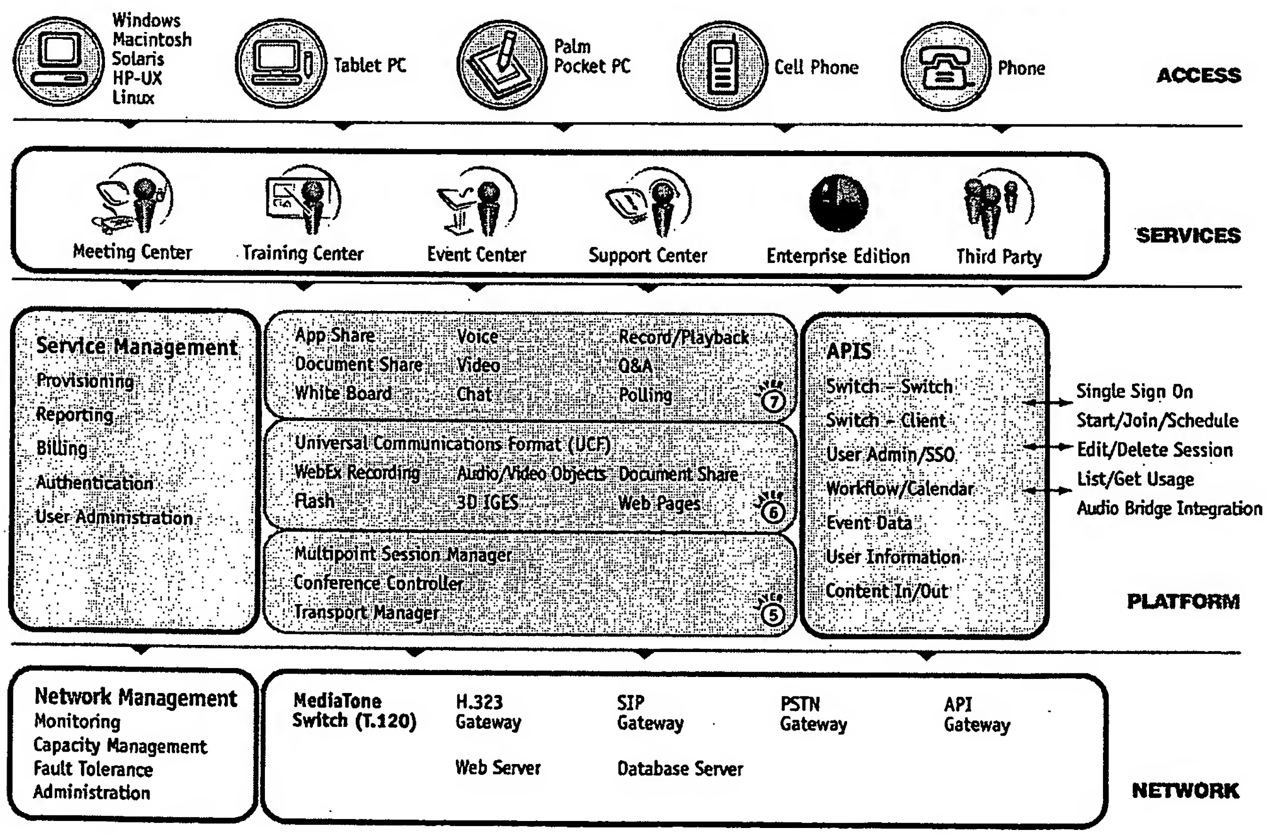
The MediaTone Platform, a distributed software architecture for Web-based communications services, is deployed over the MediaTone Network. The MediaTone Platform supports the full range of data, voice and video communications needed to provide a setting that simulates the full spontaneity and productivity of face-to-face meetings. With the MediaTone Platform, WebEx can rapidly develop new interactive Web communications services.

With the WebEx MediaTone Network:

- Latency and interruptions in multi-point interactive meetings are eliminated, even when participants are located in different countries.
- Participants may use any telephone for the audio portion of their meeting.

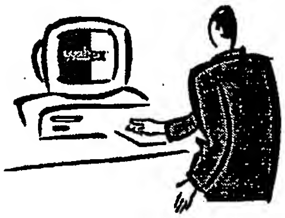
The MediaTone Platform Provides:

- Administrative capabilities, such as scheduling, provisioning and billing.
- Capabilities for session management, conference control and communications.
- Rich features, such as application sharing, video, white boarding and VoIP.



WebEx MediaTone Architecture

A More Robust T.120



The MediaTone Platform leverages the T.120 standard, which supports platform-independent, multi-point data communications. WebEx has built upon the T.120 standard, adapting it to a Web-native infrastructure and significantly enhancing the T.120 Presentation and Application Layers. WebEx also has extended the T.120 protocols for scalability, fault tolerance, security and manageability—while assuring PSTN integration. By basing the MediaTone Platform on an enhanced version of the T.120 standard and by creating a set of integration toolkits, WebEx has created a highly scalable and open information-switching network.

Originally developed by leading telecommunications providers to promote Integrated Services Digital Network (ISDN) service, T.120 is the first well-defined switched architecture for real-time data communications. The standard addresses multimedia-technology issues with attention to both voice and data requirements. The T.120 protocol, which focuses on Layers 5-7 of the OSI Model, addresses the following issues:

- Communications/transport interface (e.g., TCP/UDP).
- Multipoint session management and conference control.
- Application Layers standards.
- White boarding, application sharing and file transfer.

WebEx enhancements to the T.120 protocol suite include:

- Well-defined support for HTTP.
- Vector-based graphic format for sharing any document/format and session archiving.
- Format and protocol for sharing and synchronizing rich media.
- Federated Switched Network for global deployment and ubiquitous access.
- PSTN integration.
- A rich set of APIs and tools for integration with Web and desktop applications.
- Enhanced network scalability, reliability and manageability.
- Improved security.
- Business and operational tuning.

Service Management Layers

WebEx service management layers provide customers with an extensive array of features to maximize the use of WebEx services. WebEx service management capabilities include:

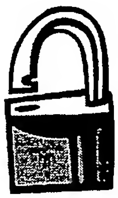
- Administration.
- Reporting and Monitoring.
- Fault/Recovery.
- Provisioning and Billing.
- Authentication.

The OSI Model

The OSI communications model defines how messages should be transmitted between any two points in a telecommunication network. The reference model defines seven layers of functions that take place at each end of a communication. Within the OSI Model:

- The ***Application Layer*** represents the level at which applications access network services, such as software for file transfers, database access and electronic mail.
- The ***Presentation Layer*** translates data from the Application Layer and manages security issues by providing services such as data encryption and compression.
- The ***Session Layer*** enables two applications on different computers to establish, use and end a session.
- The ***Transport Layer*** handles error recognition and recovery. It also repackages long messages, when necessary, and sends receipt acknowledgments.
- The ***Network Layer*** addresses messages and translates logical addresses and names into physical addresses. It also determines the routes and manages traffic problems.
- The ***Data Link Layer*** packages raw bits from the Physical Layer into frames— logical, structured packets for data.
- The ***Physical Layer*** transmits bits from one computer to another and regulates the transmission of a stream of bits over a physical medium.

A Secure Network-Friendly Solution



WebEx patent pending technologies provide universal access to the MediaTone servers from the Internet. WebEx provides a secure location on the Internet where users can connect and collaborate at any time from any place without requiring modifications to their security infrastructure.

WebEx provides encryption of all session content with Secure Sockets Layer (SSL) technology to ensure the high level of security required for enterprise data communications. The T.120 based MediaTone architecture ensures that session contents are switched through the MediaTone Network and never stored in the WebEx infrastructure.

Additional Key Capabilities of the MediaTone Platform

WebEx Universal Communications Format (UCF)

- UCF is a part of the MediaTone technology developed by WebEx that makes interactive communication powerful and effective by delivering unprecedented levels of interactivity and support for advanced multimedia communications.
- UCF technology includes a portable document format for sharing and annotating on any document, and a protocol for sharing multimedia content.
- UCF enables the ability to:
  - Share PPT presentations with animations and transitions.
  - Spontaneously show rich media in WebEx meetings.
  - Easily create engaging, interactive presentations containing rich media.
  - Share Flash animation, video, audio, Web pages, and WebEx recordings in WebEx meetings with full control over delivery.

WebEx Access Anywhere™
The WebEx Access Anywhere service enables any meeting participant to securely access or share information or applications that reside on an unattended remote computer. Mobile employees can access information on their office computers from a personal digital assistant (PDA) and share it in a WebEx meeting.

Video Conferencing
WebEx technology supports video conferencing with a browser and Webcams. WebEx also supports feeds from video technologies, including Polycom cameras and standard video camcorders. MediaTone also provides support for multi-point video conferencing.

The WebEx Solution

WebEx has developed a comprehensive set of Web communications services built on its patented MediaTone technology in order to meet the diverse requirements of its enterprise customers, enterprise portal providers, application developers and telecommunications providers. All WebEx services support highly interactive data, voice and video communications across multiple client platforms:

- Windows 95, 98, ME, NT, 2000 and XP, Tablet PC
- Macintosh OS 9 and OS X
- Solaris 7 and 8, HP-UX 11.3, Linux
- Palm OS, Pocket PC

"WebEx Event Center and Meeting Center services enable VeriSign to present important new product information to our affiliates more effectively and efficiently. Our professional services group has increased productivity by providing quality support to our customers around the globe in far less time. WebEx provides the collaboration and presentation power required to fully support our outsourced service."

Stephen Fridakis, director of professional services, VeriSign

"With WebEx we're able to train hundreds of people at any given point in time. The employee never has to leave the branch. So the branch does not lose the productivity of the employee. And we wind up with a better-trained employee. We've estimated that this is saving us more than $4 million dollars per year."

Ron Schneider, First Vice President of training and performance development, Countrywide Wholesale Lending Division

Integrating WebEx Services with Enterprise Applications

Customers can integrate WebEx services with their enterprise applications by using the WebEx integration technologies. These include APIs and software development kits (SDKs) that support integration of real-time interactive capabilities with applications such as enterprise portals, project management services, content management solutions, and learning management systems using the popular Web programming languages HTML/XML. More than 350 companies and partners have integrated their applications with the MediaTone Platform.

The WebEx APIs, together with MediaTone's support for voice/data standards and protocols, provide many benefits for corporate customers and end users of WebEx services. Upon integrating their applications with WebEx services, corporations gain powerful Web collaboration capabilities that drive productivity and reduce travel costs. In addition, the ability to integrate WebEx services with corporate accounting, CRM, ERP, human resources and other key applications streamlines maintenance and upkeep. For end users, participating in WebEx online meetings is easy and efficient; single sign on integration eliminates the need for multiple passwords, and the user can easily move between applications or documents, as the meeting requires.

WebEx Integration Capabilities

The WebEx APIs and SDKs enable the following integration capabilities for enterprise customers and teleconferencing service providers:

To manage user data:

- Sign up new user
- Login/logout
- Activate/deactivate users
- Edit users

To manage meeting scheduling and registration:

- Start/schedule/host/join
- Edit/delete
- List/add/delete attendees
- Create/get registration form
- Register attendee

To manage and access the history of online sessions:

- List/get usage history
- List recorded access history

Protocol Support

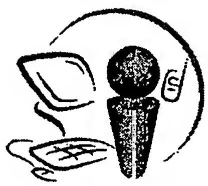
MediaTone technology supports many multimedia protocols to fulfill its mission of delivering rich-media contents to the broadest possible selection of customer devices. Supported standards and protocols include:

- H.323, the leading protocol for VoIP.
- Session Initiation Protocol (SIP), which is used for conferencing via IP phone or instant messenger (IM) device.
- Lightweight Directory Access Protocol (LDAP), a vendor-independent network directory protocol that provides directory-server integration for WebEx services.
- Extensible Markup Language (XML).
- SSL.
- Aviation Industry CBT Committee (AICC) protocol.
- SCORM.

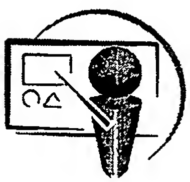
In addition, WebEx supports the Universal Communication Format™ (UCF)—a revolutionary delivery format that enables high-speed sharing and delivery of rich content as part of a presentation. UCF enables users to share content within Microsoft™ PowerPoint™ presentations, with full control over the delivery. Users can share full PowerPoint animation and transitions, just as they would in an in-person meeting, and participants may start, stop or pause the streaming content whenever they desire.

WebEx Services

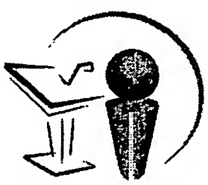
WebEx Enterprise Edition™ enables administrators to provide, and users to access, the complete suite of WebEx interactive services, leveraging the features of the specific service that's best for their needs. WebEx Enterprise Edition is delivered with My WebEx, the personal interface into the WebEx suite of services. Through My WebEx, users within the enterprise can access all of their WebEx services in a single place, with a single login.



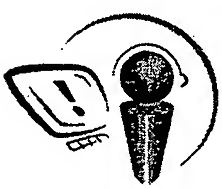
WebEx Meeting Center™ is the most powerful online Web meeting solution available. Users can present, collaborate, demonstrate, sell – anything that can be done in a face-to-face meeting can be done in a WebEx meeting. MediaTone enables productive, online meetings with rich media, using any desktop, laptop, or wireless handheld device.



WebEx Training Center™ empowers users to deliver a rich and compelling live, online classroom experience. Users can maximize the reach, timeliness, and effectiveness of training programs while decreasing delivery costs.



WebEx Event Center™ makes highly effective seminars and exciting multimedia events available with a browser, managed from a user's desktop, all for a fraction of the cost of a traditional enterprise-wide event.



WebEx Support Center™ utilizes the power of WebEx to create a perfect environment for delivering high-quality, low-cost customer support.

Benefits of WebEx Services

Benefits of WebEx services include the ability to share multiple documents and presentations in a meeting and use streaming or local multi-media content. Multiple presenters may collaborate in one meeting, and multi-language meetings are available through a customizable interface. Presenters may run any software application for effective demos, training and team meetings, including applications for:

- Customer relationship management (CRM)
- Enterprise resource planning (ERP)
- Financial management

The following capabilities are included in some or all of the WebEx services:

- Application Sharing
- Remote Control
- White Boarding
- Polling
- Chat
- Q&A
- Breakout Sessions
- Integrated VoIP
- Integrated audio conferencing

Requirements of a Web Communications Platform

Much like the telecommunications infrastructure is built on standards to support voice communications from myriads of endpoints, a Web communications infrastructure must provide for ubiquitous access, allow for an ever-increasing range of functionality, be scalable and extensible, and most importantly be reliable. The Web has been a powerful medium for ubiquitous access to information and services, and a successful communications platform should leverage it for access to its services by individuals who are anywhere in the world and who use any combination of wired or wireless Internet devices. Finally, the infrastructure should provide support for rapid provisioning of an increasing range of services with relatively low cost of ownership.

WebEx Delivers Web Communications Solutions

To meet demand for Web communications services such as online meetings, conferencing, learning and customer support, WebEx is adding live interaction capabilities to the Web. This requires strong support for service management, multi-point conference control as well as robust communications support. Additionally, with full support for sharing any documents including streaming audio and video, Flash, and other rich media formats, WebEx communications services enable users to share applications, take remote control, and interactively annotate on-screen materials. With these services, enterprises have:

- Richer, more effective online communications.
- Faster time to market and quicker problem resolution.
- Lower costs for expenses such as travel, meeting venues and meeting planning.
- Better decision-making.
- Improved productivity and efficiency through the use of real-time interactive Web communications across the enterprise.

Join hundreds of Fortune 1000 companies already using WebEx.
For more information, visit www.webex.com or contact a service consultant at +1.877.50.WEBEX (or +1.408.435.7048).

webex

011003.MTWP

WebEx History

- Founded, 1996.
- Initial Public Offering, 1999.
- Acquired and developed key technologies in real-time collaboration.
- Today offers a breadth of services that are unparalleled in the industry—plus outstanding scalability, reliability and cost performance.
- Headquartered in San Jose, Calif., with offices in Sacramento, New York, Amsterdam, Melbourne, Hong Kong and Tokyo.
- Approximately 950 employees worldwide.
- Research and Development Centers of Excellence in San Jose, Hangzhou, Suzhou and Hefei.
- Data Centers in San Jose, Denver, Virginia, Hong Kong, Tokyo, Stockholm and London.

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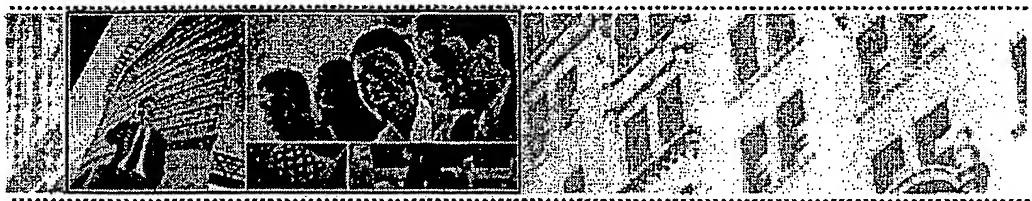

What if you could:

- Never miss an important call or fax
- Have one, unified voicemail box
- Avoid unwanted calls while only taking the important calls through sophisticated call screening features
- Set up conference calls in less than a minute via the Web or phone
- Give out only one phone number instead of 4 or 5
- Be able to save your voicemails to your hard drive

Let's be honest, you probably have way too many phone numbers. You have your office, home, cellular, fax and maybe even pager number to manage. And remember, you aren't the only one who has to manage all those numbers, you expect your customers, colleagues and family to know which number to call you on and when. With AccessLine SmartNumber, you only give out ONE NUMBER. You then decide which phone to send your callers to. You can also screen calls to make sure you take the calls you want and skip the ones you don't! If you can't take the call, your SmartNumber will take a message.

Also, your SmartNumber is your fax number. SmartNumber automatically knows the difference between someone calling you and an incoming fax. It will store your faxes and let you send them to the fax machine of your choice, or just view them via the Internet.

Simply Log onto AccessLine.com and you can route all your calls to any phone, anywhere, any time. You can view faxes and listen to voicemails right from your web browser and forward them just like e-mails!

In addition to SmartNumber you can also buy other AccessLine products by calling 877-716-2540. Or to try your own AccessLine SmartNumber immediately, click **here to sign up now!**

For more information, please contact:

Justin Bowers, Director of Small Business Sales
jbowers@accessline.com
877-716-2540

"Accessline Comms' Accessline Service, The One-Number Wonder," CommWeb, T. Kramer, February 1, 2000,http://www.cconvergence.com/article/TCM20000504S0014

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Accessline Comms' Accessline Service

The One-number Wonder.

By Travis Kramer

CommWeb

02/01/2000, 12:00 AM ET

We don't know if this is the direction in which business telecommunications are headed for everyone, or if it's a new trend for itinerant businessmen only, but lately we've been deluged with offers to try out "one-number" services. Accessline Communications (Bellevue, WA - 877-800-0999) approached us with its customizable voice and communications service, AccessLine, an all-in-one number we found perfect for any professional who needs to be in contact with colleagues, clients, and family wherever he goes (and doesn't have a good office voicemail). Accessline will store voice and fax messages, route calls to your current location, host audioconferences, and much more.

THE SETUP

Accessline is easily programmed using a touchtone keypad, and once user-initialized, it's reprogrammable by keypad or the online GUI. You're given a toll-free number and a temporary PIN when you sign up. Since setup for these types of services sometimes entails a delay (a pain when you need to change numbers in a hurry), I set up my Accessline by phone while I logged onto the Web site, and I saw the entered information immediately updated. Accessline (characterized by a friendly female voice) prompts you for your temporary PIN before you record your name and set your own PIN. Once that's accomplished, it's ready for use. When you need to tell Accessline where to forward your calls, you can set your forwarding numbers by phone or online at any time. We were given a handy card containing all the prompt codes we'd need for operating Accessline. The forwarding feature uses two-digit codes to represent the various contact numbers for easy redirection when you call in to change your forwarding options. For example, home is 10, office is 20, cellular is 30, etc. If you're

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going from home to your office, simply call your Accessline number, enter your PIN, press "2" to direct your calls, and press "20." Two great features for directing calls are the Timer and the Weekly Schedule. If you know how long you'll be near a certain phone, or your schedule demands that you be in the Pennsylvania office every Thursday from 9am til noon, for example, you can set the Accessline timer for the length of time you would like your calls sent to each forwarding number. Enter "*" if you want calls forwarded until further notice, or enter the amount of time (in minutes) that you will be available; then enter the memory number of your next location. Accessline automatically redirects the calls for you. Similarly, the Weekly Schedule automatically directs calls based on your daily or weekly activities, which you enter via the Web. Now your work calls can follow you home on weekends. Wheee!

I'VE GOT MAIL! (VOICE and FAX)

Checking your voice and fax inboxes is a cinch by phone or computer. Voicemail by phone is pretty standard; use the number keys to listen, delete, save, skip, and leave messages. I liked the deletion confirmation feature. After deleting a message, you are prompted to press "5" to confirm deletion or any other key to cancel the action. Superb for countering the caffeine twitchies. We love voicemail features that help you return messages without having to log into and out of your voicemail box; saves time, money, and aggravation. Instant Callback lets you return the call (to any caller who attaches a phone number to a voicemail message) by pressing "9*." The Rebound feature returns you to your voicemail inbox when the call is completed. You also have the option to listen to your voicemail online as a .VOX file (which stores the voice file digitally), save, delete, or even forward messages as email attachments. When checking faxes over the phone, you'll hear the number of new faxes and your delivery options. You can forward faxes in groups (new or delivered) or individually; or choose to send them to your default fax or to an alternate number. Each fax I had delivered arrived almost instantly and came with a cover sheet detailing the date and time received and the sender's fax number. If you're not near a fax machine, you can also view faxes online as .JPG or .TIFF files, or forward them as email attachments. The GUI is convenient and straightforward, working in real time with the touchtone menu.

BUT WAIT, THERE'S MORE...

Accessline offers pager notification for incoming calls, voicemail, and faxes. The page for new voice or fax messages displays the appropriate function code, followed by the number of new messages and the calling/faxing party's number. When you get a page for an incoming call on hold, go to any touchtone phone, call your Accessline number, and press "4" to be connected with the call. Call screening is great for hectic days or receiving calls at home; it screens calls based on caller discretion (asks if call is urgent) or your discretion. Accessline also lets you host an inbound conference call using your personal number as the dial-in bridge. You can schedule a conference call (date, time, duration) by phone or online. Participants call Accessline to connect to the conference. Accessline is also testing its Beta release working of an email feature that will configure your POP3 email accounts with your Accessline inbox. Basic service is $18.95/month plus cost of calls (10/min.) and a $40 activation fee.

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"InteleScreener," 2003,
http://www.intelescreener.com/howitworks.html

InTeleScreener™

A Revolution in Telephone Privacy

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- Operating Manual
- FAQs
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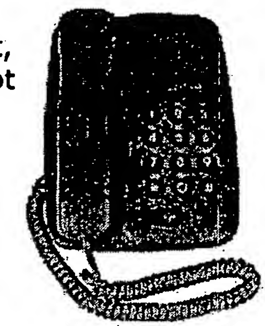
Company Information

The **InTeleScreener™** works with your telephone company's Caller ID service. Caller ID information is sent between the first and second ring on an incoming phone call. The **InTeleScreener™** blocks the first ring from ringing the telephone and processes the Caller ID information to screen the phone call prior to the second ring. Depending on the mode of operation the call will either be blocked or allowed to ring your telephone.

Block Mode



During block mode, any incoming calls from a number that is in the block list, the **InTeleScreener™** will block them from ringing the telephone. Callers not in the block list will be allowed through to ring your phone. Numbers can be added to the block list simply by pressing BLOCK while viewing the Caller ID display or later while reviewing the history list.

The **InTeleScreener™** has the two following options for blocking callers in the block list, both of which never allow your telephone to ring:

Block Mode: Disconnect Call
This option will answer the call, emit the disconnect (SIT) tones to the caller making your phone look like it has been disconnected and then hang-up. This is all done automatically by the **InTeleScreener™** without allowing the telephone to ring and disrupt you.

Block Mode: Do Not Answer
This option will prevent your phone from ringing, but will not disconnect them or emit the disconnect tones. To the caller it will appear as if your phone is ringing and you are not answering. Use this option when you do not want to disconnect the caller or want to allow caller to leave a voice message. It is also useful when you have multiple **InTeleScreener™** devices and wish to block a phone call in one room while allowing the call to ring another telephone in a separate room. This mode is useful for teenagers or if you live with roommates and do not want people

calling for them to ring your phone.

▶ Screen Mode

In Screen mode your phone will only ring when the call is from someone in the Screen List such as family and friends. Screen mode allows you to choose who can ring your phone. Numbers can be added to the screen list simply by pressing SCREEN while viewing the Caller ID display or later while reviewing the history list. No more wrong numbers in the middle of the night or telemarketers bothering you during a quiet dinner.

▶ Auto Mode

The **InTeleScreener™** is fully programmable so the user can customize it to his/her lifestyle. Auto mode allows the user to program the **InTeleScreener™** to run Block and/or Screen mode during preset time periods throughout the day or night. Whether you sleep during the day or night you will be able to sleep peacefully while the **InTeleScreener™** screens all of your phone calls.

Consumers more than ever are wanting to stop telemarketers from calling and disrupting their privacy. Traditional call block services from telephone companies used to block telemarketing calls are costly and ineffective at stopping telemarketers from calling. Many phone companies offer anonymous phone call screening services to stop telemarketers and unknown callers requiring them to un-block caller ID or identify themselves before they can ring through. The InTeleScreener was invented not only to block telemarketers but to give the consumer the best available product ever for stopping telemarketers from calling and all other unwanted callers from ever ringing their phone. If you want the best Caller ID device to block telemarketers and you want unbeatable phone call screening to finally stop telemarketers, you need to get the InTeleScreener Caller ID today. Stop paying for expensive call block services from your phone company and get the InTeleScreener to finally stop telemarketers from calling and disrupting your privacy.

Order online today or call us **toll free** ***1-866-884-9524***

"TeleZapper from Privacy Technologies," Privacy Corps - Our Review, 2002,http://www.privacycorps.com/pages/product1.htm

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Fighting For Your Privacy

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STARTLING STATISTICS

Charities make more money from *selling your name and number* to the other telemarketing companies than from the donations they collect from calling.

REFER A FRIEND to Privacy Corps

Your Email:

Their Email:

send

TeleZapper from Privacy Technologies

Our Review

Under real test conditions, on average, the number of telemarketing calls per test site was reduced from two telemarketing calls a day to approximately two calls per month, over a period of four weeks. We found these test results pretty darned good!

The TeleZapper was incredibly easy to connect using the simple diagrams provided. The telephone cords connect in two places, one into the TeleZapper from the telephone-company line and one out of the TeleZapper to your telephone. All you have to do is unplug the line to your phone and connect the TeleZapper in its place. A connection cord is also provided to connect the TeleZapper to the wall jack. You'll need to plug the Telezpapper into an outlet to power the TeleZapper unit, but the manufacturers have provided a plug and cord.

Once connected, the TeleZapper works in the background and requires no further input from the user. You'll only need one TeleZapper for all phones connected to that line. However, if you have more than one telephone number, you'll need a TeleZapper for each additional phone line.

The tone transmitted when the telephone is answered is very short and subtle, and testers report that most callers don't even hear it. If they do notice it, generally the comments came from callers after their third or forth call, but most didn't feel that it was at all intrusive.

We also tested the product with the company phone system which has it's own control unit, intercom, and multiple phones. The TeleZapper worked just as well under these conditions, although because our business uses several lines, we needed several additional TeleZappers.

As an added benefit, the TeleZapper continues to work in the background even when

an answering machine picks up the phone, resulting in a diminished number of calls over time. The calls that did get past the TeleZapper were manually dialed calls from smaller, more local companies, dialing numbers for each area in sequence.

Privacy Corps recommends this product as a good value, in terms of ease of use, effectiveness, and reliability. We were also pleasantly surprised by the free technical support available from the manufacturer via their toll-free number.

Purchase Now!

Customer Testimonials

Just wanted to let you know I bought the TeleZapper out of desperation a few weeks ago. My home telemarketing calls have now STOPPED!!! THANK YOU SO MUCH! You have saved my sanity. *Sue - Manassas, VA*

Thank you, TeleZapper! You've saved me the cost of replacing the glass in my bedroom window, and a doctor bill for trying to throw my phone a hundred yards! Yes, I'm talking about those late, and I do mean late, nighttime calls trying to sell me everyting from newspapers, family portraits, to life insurance! Ah, privacy...what a beautiful word! *Ruben - Palmdale, CA*

We would love to hear from you if you have used one of our products!

CLICK HERE to fill out our product feedback form.

Frequently Asked Questions (FAQ)

- How do telemarketing calls work?
- How does my number get on telemarketing lists?
- How does the TeleZapper "zap" telemarketers?
- Will the TeleZapper "zap" calls from anyone other than telemarketers?

How do telemarketing calls work?

There are several hundred telemarketing call centers in the U.S., with the vast majority of telemarketing calls being dialed by a computer known as an auto dialer or predictive dialer. Predictive dialers can dial 3-5 numbers simultaneously and can make as many as 500,000 calls between 8 a.m. and 9 p.m. When you answer your phone, the computer connects you to a live telemarketer who tries to sell you something. If you are not home or if the computer gets your answering machine, your number will be put back in the database to be called again later.

Back to top...

How does my number get on telemarketing lists?

You can get on telemarketing lists in many ways:

- By having a listed telephone number;
- Through a reverse phone book organized by neighborhood;
- When you dial an 800 number that uses an Automatic Number Identification system (ANI) to record your number;
- Via credit information services, such as Equifax, etc.;

- By ordering products or services from direct marketers or catalogs, whether you order through the mail, from web sites, or via 800 numbers;
- By printing or including your telephone number on your personal checks;
- Even by simply paying your monthly bills.

These lists of telephone numbers are then often sold, bartered, rented, shared and copied from one telemarketer to another. As your number constantly finds its way onto new call lists, the TeleZapper will continue to do its job over time to help you protect your privacy.

Back to top...

How does the TeleZapper "zap" telemarketers?

The TeleZapper uses the technology of telemarketers' automatic dialing equipment against them. When you or your answering machine picks up a call, the TeleZapper emits a special tone that "fools" the computer into thinking your number is disconnected. When the computer hears the tone, it hangs up before you can be connected to a telemarketer and then deletes your phone number from its database. Overtime, as your number is removed from more and more databases, you'll see a dramatic decrease in the number of annoying telemarketing calls you receive.

Back to top...

Will the TeleZapper "zap" calls from anyone other than telemarketers?

The TeleZapper is designed to "zap" calls made by predictive dialer computers by doing two things: first, by disconnecting predictive-dialed calls before you can be connected to a live telemarketer and second, by deleting your phone number from telemarketing computer lists. Whether the TeleZapper will affect computer-dialed calls from other sources depends on the type of computer equipment and how that equipment is being used. Therefore, it may also "zap" calls from other organizations that use predictive dialer computers, such as charitable organizations, blood banks, public safety and service organizations, market researchers, opinion and political pollsters, and academic institutions.

Many organizations and communities do not rely entirely on computerized calling systems to reach you. Most have secondary means in place to contact or notify people with important information. Furthermore, these organizations can always contact you by simply dialing your phone number manually. Manually dialed calls will not be zapped. As such, you can notify organizations to determine if they use predictive dialers and, if so, to ask that your phone number be manually dialed or that alternate means be used in order to contact you.

Finally, during times of severe weather or at any time that important public emergency notifications might be received, you can quickly and easily disconnect your TeleZapper to allow all calls, including those placed by computerized dialers, to be successfully completed.

Back to top...

How do I know when I've "zapped" someone?

If you answer your telephone and there's no one there, the odds are that you just "zapped" a telemarketer. After a few weeks, you'll notice that you are receiving fewer and fewer of these calls.

Back to top...

Why buy a TeleZapper instead of letting my answering machine or caller ID screen telemarketing calls?

The TeleZapper is the only product that emits a signal that "tells" predictive dialer computers your number is disconnected. Unlike answering machines or caller ID, once the TeleZapper's tone is emitted, your number is removed from the computer's call list. So, as time passes, you'll receive fewer and fewer annoying telemarketing calls. If the computer gets your answering machine, your number is put back into the database to be called again and again ... and again. Most telemarketing calls show up on Caller ID as "out of area" or "private". But since many callers are identified in these ways, it's difficult to know who's calling and whether or not you want to pick up the phone. The TeleZapper really is a better solution to keep telemarketers out!

Back to top...

How do I install the TeleZapper?

The TeleZapper is easy to install. Simply plug it into any electrical outlet and phone jack to cover all extensions and answering machines connected to that line. Plus, you have flexibility to place the TeleZapper anywhere in your home since you can plug the phone cord into a phone, into an answering machine, or directly into a phone jack.

Back to top...

Do I need a TeleZapper for each telephone extension?

No. One TeleZapper covers all telephones and answering machines connected to the same line (telephone number). If you have two lines, you will need an additional TeleZapper for your other line.

Back to top...

Do I need to use a special phone with the TeleZapper?

No. The TeleZapper works on any home phone line and with any type of phone.

Back to top...

Will the TeleZapper interfere with my answering machine?

No. The TeleZapper will not interfere with your answering machine. In fact, the TeleZapper works with your answering machine to "zap" telemarketers when you are away or when you prefer not to answer the phone. When your answering machine picks up a call for you, the TeleZapper emits its special tone to "zap" the telemarketer. Your name will have been deleted from another telemarketing list and you won't have been bothered at all. We do recommend that you re-record your message and delay speaking for a few seconds to allow time for the TeleZapper tone prior to the start of your recorded message. A caller who wishes to leave a message on your answering machine will hear a short tone followed by your recording.

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I have voicemail from the telephone company. Will the TeleZapper work with this?

Yes and No. Your telephone must go "off-hook" for the TeleZapper to emit its tone. As long as you pick up a call, the phone goes "off-hook" and the TeleZapper emits its tone to "zap" telemarketers. If, instead, the telephone company "answers" your calls through voicemail, your phone does not go "off-hook" and the TeleZapper cannot emit its tone. The TeleZapper will not interfere with the normal operation of your voicemail.

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Will the TeleZapper interfere with my computer or fax machine?

No. The TeleZapper does not interfere with the operation of your computer, your fax machine or other telecommunications or electronics equipment.

Back to top...

Will the TeleZapper work on junk faxes?

Yes. If your fax number is dialed by a computerized predictive dialing system that is programmed to listen for disconnected numbers, the TeleZapper will "zap" your fax number from those calling lists.

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When is/should the red LED light on the TeleZapper be on?

The red LED light flashes three times when the phone is picked up and the TeleZapper emits its tone. Since the light is off at all other times, it does not mean your TeleZapper is not working when the light is not illuminated.

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What's to keep telemarketers from just turning off the "disconnected" feature on their computerized dialing equipment?

Plenty! There are millions of non-working telephone numbers. Telemarketers succeed by efficiently connecting their operators to PEOPLE and then selling them something. The last thing they want is to waste time being connected to a non-working telephone number. Plus, when you've installed a Telezapper, you're telling the telemarketer you do not want to talk to them. There are laws that support your right to privacy and most telemarketers really don't want to violate those laws -- they just want to talk to someone who might buy something.

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Does the TeleZapper block collectors' computer generated calls made to collect legitimate debts?

By purchasing the TeleZapper, you are making a choice to protect your privacy. Before the introduction of the TeleZapper, you may have used other methods to screen unwanted calls, such as Caller ID, Privacy Manager, and answering machines. The TeleZapper is simply another option for people who are concerned about maintaining their privacy. That right is a fundamental one and is, indeed, constitutionally protected.

The vast majority of telemarketing calls are computer dialed at random. The TeleZapper is designed to "zap" your phone number off telemarketers' computers. Whether the TeleZapper will affect computer-dialed calls from other sources, such as collection agencies, depends on the type of computer equipment being used and how that equipment is being used. Therefore, the TeleZapper may also "zap" calls from organizations, other than telemarketers, that also use predictive dialer computers.

The TeleZapper will not "zap" bill collectors or other companies who dial a telephone number manually rather than through a predictive dialing computer.

Back to top...

Does the TeleZapper block incoming calling card calls?

Telezapper has learned that some long distance calling card manufacturers use a system similar to predictive dialing systems. Whether the TeleZapper will affect computer dialed calling card calls depends on the type of computer equipment and how that equipment is being used. Since the TeleZapper is designed to disconnect calls before a computerized predictive dialer can connect you to a live telemarketer/caller, there is also a chance that predictive-dialed calling card calls may not be completed.

Back to top...

Does the TeleZapper create a computer virus for the caller?

The TeleZapper DOES NOT create a computer virus for callers. The TeleZapper simply emits a tone that predictive dialer computers recognize as a disconnected number.

Back to top...

Does the TeleZapper interfere with a DSL connection?

The TeleZapper DOES NOT interfere with a DSL connection. Be sure that the Telezapper is connected at a point after the DSL junction box.

Back to top...

What are the differences between the TeleZappper and TeleZapper II?

Besides being smaller in size the TeleZapper II offers a user-selectable switch to choose between emitting one or three tones. It is also powered by a lithium battery with a projected useful life of approximately 7 years, and does not require household current.

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"A Proposal for Internet Call Waiting Service Using SIP," A. Brusilovsky et al., Lucent Technologies, PINT Working Group, Internet Draft, January 1999.

PINT Working Group
Internet Draft

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Expires: July 1999

A Proposal for Internet Call Waiting Service using SIP
An Implementation Report

<draft-brusilovsky-icw-00.txt>

Status of this Memo

Abstract

The purpose of this Internet Draft is to start discussion on the issues involved in Internet Call Waiting Service (ICW), as part of interconnecting IP and Global Switched Telephone Network (GSTN) with the intent of providing ICW service that is much needed by numerous dial-up Internet users. Interworking of the IP network and GSTN, based on open well-defined protocols, will promote interoperability of both the networks and systems built by different vendors. This Internet Draft is submitted with the goal of becoming an informational RFC.

The rest of this document is as follows:

Section 2 briefly describes the services offered to the end Subscriber. It is the support of these services that necessitates the proposed internetworking project.

Section 3 describes the scope of the proposed project by introducing its overall architecture, identifying the interfaces to be standardized, describing experience with SIP for ICW.

Sections 4, 5, and 6 respectively address security considerations, supply references, and provide the authors address, as required by [1].

Section 7 acknowledges individuals providing assistance in the creation of this document.

Section 8 is the Appendix, which contains IN Tutorial and Figure A.

2. Service Description

It is a well-known problem that call waiting tone interferes with the operation of a modem. Anyone using the telephone for a modem connection to a host computer can not gracefully deal with incoming call waiting calls. Internet Call Waiting is the capability to provide incoming call notification and completion options when the Subscriber is on a dial-up IP connection. When a call comes in the Subscriber is presented with a pop-up dialog box, that presents the caller's number and, optionally, his or her name. Internet Call Waiting solution provides a simple, graphical-oriented way to notify subscribers while connected to the Internet, of incoming calls. It allows the subscriber to accept or reject the call.

Benefits

Service providers can achieve the following important benefits through the use of Internet Call Waiting Service:

o More calls completed. Call completion is an important aspect of the service provided by telecommunication operators. Calls that end in busy or no- answer, consume network resources. Solution like Internet Call Waiting contributes to greater call completion which lowers expense and provides value to both the consumer and service provider.

o The ICW platform is the foundation to offer services: The service provider has the opportunity to enhance Internet Call Waiting with other services like Internet Follow-me, personalized call management, unified messaging service, click to return (dial) an important call, and other call management functions which integrate voice and data services.

o Service provider can offer the following important benefits to the subscribers through the use Internet Call Waiting Service:

- Simple way to manage voice and data calls over a single telephone line.
- Ability to track all incoming calls while the service is active
- PC Graphical Subscriber Interface provides a simple intuitive Subscriber interface and also allows easy customization.

3. Scope of the Proposed Project

Figure A illustrates the hardware architecture that will support ICW Service. The lines indicate the control and/or voice paths. Control paths are labeled by the protocol that will be used over them.
IN elements (SCP, SMS, SSP) are specialized servers, connected to switches and other network elements. They handle data queries and updates, specialized call routing and other advanced telecom services. For more information on Intelligent Network please see our IN Tutorial in the Appendix of this Internet Draft.

The following software components make up the ICW architecture.

o ICW User Agent Server (UAS) - The ICW UAS (SIP Client) and server communicate via the SIP protocol over TCP/IP. The ICW UAS can start up automatically as soon as a PPP connection is established. It also responds to the incoming request for call treatment by popping up the dialog box to the subscriber presenting information about the Calling Party and asking for an Accept or Reject decision. The UAS sends the resulting choice back to the ICW server. In the case of a accepted call, the UAS drops the modem connection to the ISP to allow the incoming call to complete.

o ICW server - a SIP proxy server that perform the following

functions. The SCP is not being used as a general-purpose database host. Thus, SIP-related database dips are envisioned to be in the domain of a generic ICW server which can interface with any commercial-grade database engine or any LDAP-enabled database. The SCP is free to provide telecommunication intensive tasks that it was designed for.

- Listening for incoming messages from the application running on SCP
- Providing a data store mechanism for ICW applications
- Handling Web-based GUI (Applet) requests for subscriber provisioning on the ICW server

o SCP platform software - The ICW APPLICATION runs on SCP

- ICW Application runs on SCP - The AIN 0.1 Terminating Attempt Trigger (TAT) is used to enable PSTN call handling. Thus, the Application responds to an AIN message for every call to the subscriber. For each call, the Application either returns a request for normal routing, if the subscriber is no longer active,

or sends a message to the ICW server passing along the calling number. Based upon the reply from the ICW server, which may be Accepted or Rejected, the SCP sends the appropriate instructions ba ck to the SSP.

Various alternatives exist for firewall support. The ICW UAS-to-ICW server firewall could be standard corporate security firewall. However, the security policy would need to allow TCP-based SIP messages to flow between the ICW UAS and server over the standard SIP port 5060. The ICW server-to-SCP firewall is optional and could be used to provide an extra level of protection for the SCP by restricting Intranet access or by enforcing a more restrictive security policy than the outer firewall. General and ICW specific security considerations are covered in Section 4.

Other components in the diagram are part of the standard Internet and PSTN and include the Internet Service Provider (ISP), ISP modems and web servers, the Service Switching Point (SSP) and the Signal Transfer Point (STP). The SSPs must be provisioned with the necessary trigger for the ICW service, the AIN 0.1 Terminating Attempt Trigger.

When the Calling Party dials the ICW Subscriber's Destination Number, the Calling Party experiences the standard Call Waiting treatment, ringing, until Calling Party abandons or the Subscriber specifies treatment: Subscriber treatment options and Calling Party experience are:
o Refuse Call: Calling Party hears ringing until Calling Party abandons. In SIP terms, this results in the SIP UAS sending a "603 Decline" message to the ICW server.
o Hold Call: Calling Party hears [optional] announcement to hold while "other" call in progress is completed. The intent is that the Subscriber will accept the call momentarily. (Another possibility would be to tell the Calling Party that you'll call them back in a few minutes, etc) In SIP terms, this results in the SIP UAS sending a "182 Queued" message to the ICW server.
o Send to Voice Mail (assuming Subscriber has a Voice Mail service): Calling Party hears voice mail system announcements. (This redirection to voice mail could, as well, have been redirection to some other DN, e.g. cell phone, second line, secretary, etc) In SIP terms, this results in the SIP UAS sending a "380 Alternative Service" to the ICW server.
o Accept Call: Calling Party hears ringing until is connected to Subscriber. In SIP terms, this results in the SIP UAS sending a "200 OK" to the ICW server.
Note: Optional treatment options can include taking call via VoIP and route call to a third party number.

In the proposed Architecture, the Subscriber is assumed to have PPP service through their ISP. They are surfing the Internet or working at home, connected to a corporate intranet. Two components of ICW

reside on their PC; an H.323 client for VoIP and an ICW UAS to drive the presentation to the Subscriber of Setup and Notification. Controlling the ICW service is the ICW server for Internet related control and the combination of the SCP and SSP via AIN functionality providing PSTN control via SS7. There is an ICW control session between the PC and the ICW server. Controlling the VoIP aspect is the H.323 client at the PC and the H.323 gateway with H.323 packets going between them via the internet. The SCP controls the IP via Bellcore's GR-1129. The SCP and ICW server have a TCP/IP connection. The call path of the accepted call consists of the Calling Party being routed to the IP (intelligent peripheral) and bridged to the ICW Subscriber from the H.323 gateway. Firewall appliances are placed on all IP connections of the service provider. A call scenario below walks through this architecture. Integration of the H.323 GW and IP as well as the SCP and ICW server is a possibility for future enhancements.

Call Scenario

Subscription to the service.
o Subscriber signs up for the service.
o Subscriber downloads and installs the ICW UAS software.
o Subscriber Information is provisioned in the SMS (and SCP).

Activation of the service and coordination with the ICW Server (Transparent for the ICW User)
o ICW UAS establishes TCP connection.
o Subscriber authenticates himself/herself and Register with ICW Server using the encrypted password and phone number.
o ICW Server stores information in database.

Call Arrival
o Calling Party initiates call to Subscriber.
o SSP (Switch) encounters TAT.
o SCP query launched.
o SCP determines if call is for an ICW subscriber (if not then other service logic applies).
o SCP sends a SIP "INVITE" message with Calling Number, optional Calling Name and Called Number (and receives a SIP acknowledgement from the ICW Server)
o If ICW is activated for the called subscriber, ICW Server returns "TRYING" to SCP. The SCP instructs SSP to play an announcment, e.g. ringing. ICW Server determines, based on the Called Number and the IP Address of the ICW UAS and sends the SIP INVITE message to the ICW UAS.
o If ICW is not activated ICW Server returns "NOT FOUND" to SCP. SCP returns an Authorize Term message to the SSP so call proceeds as normal.

Communicating subscriber's choice to the SCP.
o ICW UAS returns a SIP "DECLINE" (for normal SSP treatment) or "OK"

(for connecting the call).
o ICW Server passes along the SIP message to the SCP

Choice: Drop Modem, take call.
o ICW UAS causes Modem to drop.
o SCP instructs switch to continue with the call (Authorize Term).
o Switch connects Calling Party to Subscriber line causing the phone

to ring.

Choice: Send to Voice Mail.
o SCP sends Authorize Term message to switch to deliver the call to the subscriber's line.
o SSP detects Busy and uses standard Call Forwarding on Busy to send to Voice Mail

Experiences in using SIP for ICW Project

The biggest advantage to using SIP in the ICW project was its ASCII-based nature and a concise set of messages. We were able to get a bare-bones SIP server running in a good part of a week. SIP is geared towards Internet protocol services; ICW is a prime example of such a service. SIP's semantics lend themselves very efficiently to the semantics of the ICW service. SIP has a very rich set of response codes that we were able to tailor to the various ICW states, such as the user accepting a call, declining a call, redirecting a call to a new location, or simply not being on the PC when the call notification arrived. Another advantage of SIP is that a SIP-based architecture is easily explained to even those who do not possess an in-depth understanding of Internet in general and IP protocols in particular. Various SIP entities like SIP User Agent Server, Proxy Server, Redirect Server, etc. lend themselves to a very extensible architecture.

The disadvantages of SIP are few; one of them being its constant state of flux. During ICW development, the SIP draft RFC changed no less then 3 minor versions. This made it somewhat difficult to agree on a standard. However, this disadvantage will be mitigated in the future when the SIP draft becomes a Draft Standard. The other big disadvantage was driven by the general lack of support for database queries. For instance, an SCP would like to authoritatively know if a user was on the Internet before sending him/her the call notification. However, the SIP message set did not support general querying capabilities for this purpose. We ended up using the SIP OPTIONS message for this purpose, even though the draft mandates that OPTIONS message is used primarily for capability set negotiations. Finally, the SIP RFCs are becoming more complex with each new revision. We believe that while adding features is critical, it would be in the best interest to maintain the simplicity of SIP for rapid development, debugging, and deployment.

Security Considerations

ICW communications between the PC and the ICW Server may travel over the Internet. Thus it is essential to provide encryption for the communications. In addition to encryption, and to make sure that the PC belongs to a registered subscriber, it is also necessary to provide authentication of both the end points; i.e. ICW Server and the PC. ICW security has been designed to authenticate both end points and if the authentication succeeded, encrypt the communications (control channel) using a symmetric key. This key is provisioned in the ICW Server database as well as generated at the subscriber's end-point (the PC) when the software is initially installed. In the future, migration of the ICW security infrastructure to SSL is envisioned.

ICW Security Requirements are, assentially, the same as PINT Security Requirements outlined in [4]:

o Peer entity authentication to allow a communicating entity to prove its identity to another in the network. Two types of peers should be recognized for the purposes of this project: end-user and the Web server, and Web server and SN. Between the end-user and Web server the authentication could be accomplished by means of the user name and password combination. In addition, encrypted communications could be used in this case. Same could be used between the Web

server and SN, but it is proposed that additional security be accomplished by replicating a part of the server's data base relevant to the business providing the service.

o Non-repudiation to account for all operations in case of doubt or dispute. This could be achieved by logging all the information pertinent to the Web transaction. In addition, the PSTN network will maintain its own account of the transaction for generating bills.

o Confidentiality to avoid disclosure of information without the permission of its owner. Although this is an essential requirement, it is not particular to the proposed project.

o End-user profile verification to verify if the end user is authorised to use a service.

In the course of the project execution, additional requirements are likely to arise and many more specific security work items are likely to be proposed and implemented.

Some of the ICW-specific security considerations:
o Hacking is a threat to any Service Provider (PSTN, Intranet, Internet). It is real danger - phone companies are common targets
o Strong Firewall solutions are needed
o Fraudulent Subscription is one of the threats
o Existing mechanisms applied to the Internet can be implemented
o Stealing a Call is a new type of security threat

o Denial of telephone service attack is possible
o Encrypted password protection can be used as one of the possible solutions.

5. References

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Glossary

AIN	Advanced Intelligent Network
API	Application Program Interface
DN	Destination Number
GSTN	Global Switched Telephone Network
ICW	Internet Call Waiting
IN	Intelligent Network
IP	Intelligent Peripheral
PSTN	Public Switched Telephone Network
POTS	Plain Old Telephone Service
SCP	Service Control Point
SIP	Session Initiation Protocol
SN	Service Node
SMS	Service Management System
TAT	Terminating Attempt Trigger
UAS	User Agent Server (SIP Terminology)
VoIP	Voice over IP (Internet Protocol)

7. Acknowledgments

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8. Appendix (IN Tutorial and Figure A)

Intelligent Network (IN), excerpt from [4]

IN ([2], [3]) is an architectural concept that provides for the real-time execution of network services and customer applications in a distributed environment consisting of interconnected computers and switching systems. Also included in the scope of IN are systems and technologies required for the creation and management of services in this distributed environment.

In PSTNs, user's telephone terminals and fax machines are connected to telephone switches. The switches (which can be Central Offices--for wireline communications and Mobile Switching Centers (MSCs)--for wireless communications) are specialized computers engineered for provision of services to the users. The switches themselves are interconnected in two ways: 1) through trunks on which the voice is carried and 2) through a specialized fault-tolerant data communications network, which is (principally) used for call setup and maintenance. This network is called (after the ITU-T standard protocol suite that it uses) Signalling System No. 7 (SS7). In addition, the switches are connected to general purpose computers that support specialized applications (called Operations Systems) whose role includes network management, administrative functions (e.g., billing), maintenance, etc. Operation systems are not connected to the switches through the SS7 network, which is, again, engineered only for set-up and real time maintenance of calls. In most cases, X.25 protocol is used for communications between operations systems and switches. Even a simple two-party call in most cases involves several switches, which may also be located in different PSTNs. To this end, the switches alone comprise a complex distributed processing environment. As far as the end users are concerned, the switches are ultimately responsible for delivering telecommunications services. Certain elementary services (such as provision of the dial tone, ringing the called line, and establishing a connection between two users) are called basic services, and all switches can presently cooperate in delivering them to end users.

In addition, a multitude of services (such as Freephone [a.k.a. 800 number in North America], Conference Calling, Call Forwarding, and many others) require much more than basic call processing. Such services are called Supplementary Services, and their implementation requires that specialized applications (called Service Logic) be developed. Developing switch-based service logic for each supplementary service would be an extremely expensive (if at all possible) task, which--in the presence of multiple switch vendors--would also require an extensive standardization effort.

The IN architecture is the alternative which, in a nutshell, postulates using a network-wide server (called Service Control Function [SCF]). The SCF executes service logic and instructs the switches on how to complete the call. A switch is involved only in executing the basic call process, which is interrupted (at standardized breakpoints called triggers) when specialized service logic needs be executed. On encountering such a breakpoint, the switch issues a query to the SCF and waits for its instruction. In addition (and this is essential for supporting the services described in section 2), the SCF may initiate a call on its own by instructing switches to establish necessary connections among themselves and to the call parties.

Physically, the SCF may be located in either stand-alone general purpose computers called Service Control Points (SCPs) or specialized pieces of equipment called Service Nodes (SNs). In addition to executing service logic, a service node can perform certain switching functions (such as bridging of calls)as well as a set of specialized functions (such as playing announcements, voice recognition and text-to-speech conversion). An important distinction between an SCP and SN is that the former is connected to switches via the SS7 network while the latter communicates with the switch via Integrated Services Digital Network (ISDN) Primary or Basic Rate Interfaces (PRI or BRI), which combine both the signaling and voice paths. With the present state of IN standardization, in principle, either an SCP or SN could be connected to an Internet server in order to support the services outlined in section two. To further narrow the scope of work so as to produce tangible results as soon as possible, the proposed project specifically addresses only interconnection between a server and SN.

Within the IN architecture, the relevant administration of the network entities (i.e., setting the triggers in the switches, transferring externally developed service logic to SCPs and SNs, and maintaining the network databases with the customer-related data) is performed by a specialize Operation System called Service Management System (SMS).

Figure A

```
                                                      |F|
                                                      |i|
+----------------------+       =============          |r|
|           +-------+|        ||          ||  SIP    |e|      +--------+
|  PC Access |  ICW  ||        || Internet |+-..-..|w|-..-..-|  ICW   |
| o..........+  UAS  ||        ||          ||        |a|      | Server |
|           +--+-----+|        =========+===         |l|      +---+----+
|  Voice Access :     |                 |            |l|          |
| O------------I:     |                 :                    ______:_____
|              I:     |                 | SIP                  Firewall
|              I:     |                 :                    -----------
|              I:     |                 |                          |
|              I:     |             +---+---+                      :
|              I:     |             |  ISP  |                      |
|ICW           I:     |             +---+---+                      :
|Subscriber    I:     |                 |        +-------+   +---+---+
+--------------I:-----+     PPP         :        |  SMS  |---|  SCP  |
               I:.......................+ |      +-------+   +---+---+
               L----------------------+: :                       s
                  POTS          =======I:=|=========              |
                             ||       I: :        ||         =====s====
                             ||       I: |        ||         ||       ||
                             ||      +++-+---+    ||         || SS7   ||
  o--------------------------+-------|  SSP  |    ||         ||Network||
Calling Party                ||      +---+---+    ||         =====s====
(Voice Access)               ||        s P S T N  ||              |
                             ========= | =========      SS7       s
                                       s-s-s-s-s-s-s-s-s-s-s-s-s+
                                                Signaling
```

Legend:
..-..- - SIP (over IP)
s-s-s   - SS7 signaling links
-----   - POTS connection
.....   - PPP connection

draft-brusilovsky-icw-00.txt>   Expires: May 1999

"A Model for Presence and Instant Messaging", M. Day, et al. Fujitsu, February 2000, Network Working Group, Request for Comments 2778.

Network Working Group
Request for Comments: 2778
Category: Informational

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February 2000

A Model for Presence and Instant Messaging

Status of this Memo

This memo provides information for the Internet community. It does not specify an Internet standard of any kind. Distribution of this memo is unlimited.

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Abstract

This document defines an abstract model for a presence and instant messaging system. It defines the various entities involved, defines terminology, and outlines the services provided by the system. The goal is to provide a common vocabulary for further work on requirements for protocols and markup for presence and instant messaging.

1. Introduction

A presence and instant messaging system allows users to subscribe to each other and be notified of changes in state, and for users to send each other short instant messages. To facilitate development of a suite of protocols to provide this service, we believe that it is valuable to first develop a model for the system. The model consists of the various entities involved, descriptions of the basic functions they provide, and most importantly, definition of a vocabulary which can be used to facilitate discussion.

We note that the purpose of this model is to be descriptive and universal: we want the model to map reasonably onto all of the systems that are informally described as presence or instant messaging systems. The model is not intended to be prescriptive or achieve interoperability: an element that appears in the model will not necessarily be an element of an interoperable protocol, and may not even be a good idea.

In this document, each element of the model appears in upper case (e.g., PRESENCE SERVICE). No term in lower case or mixed case is intended to be a term of the model.

The first part of this document is intended as an overview of the model. The overview includes diagrams, and terms are presented in an order that is intended to help the reader understand the relationship

between elements. The second part of the document is the actual definition of the model, with terms presented in alphabetical order for ease of reference.

The overview is intended to be helpful but is not definitive; it may contain inadvertent differences from the definitions in the model. For any such difference, the definition(s) in the model are taken to be correct, rather than the explanation(s) in the overview.

2. Overview

The model is intended to provide a means for understanding, comparing, and describing systems that support the services typically referred to as presence and instant messaging. It consists of a number of named entities that appear, in some form, in existing systems. No actual implementation is likely to have every entity of the model as a distinct part. Instead, there will almost always be parts of the implementation that embody two or more entities of the model. However, different implementations may combine entities in different ways.

The model defines two services: a PRESENCE SERVICE and an INSTANT MESSAGE SERVICE. The PRESENCE SERVICE serves to accept information, store it, and distribute it. The information stored is (unsurprisingly) PRESENCE INFORMATION. The INSTANT MESSAGE SERVICE serves to accept and deliver INSTANT MESSAGES to INSTANT INBOXES.

2.1 PRESENCE SERVICE

The PRESENCE SERVICE has two distinct sets of "clients" (remember, these may be combined in an implementation, but treated separately in the model). One set of clients, called PRESENTITIES, provides PRESENCE INFORMATION to be stored and distributed. The other set of clients, called WATCHERS, receives PRESENCE INFORMATION from the service.


```
         +---------------------------+
         |      PRESENCE SERVICE     |
         |                           |
         +---------------------------+
             ^                 |
             |                 |
             |                 v
      +------------+    +------------+
      | PRESENTITY |    |  WATCHER   |
      +------------+    +------------+
```

Fig. 1: Overview of Presence Service

There are two kinds of WATCHERS, called FETCHERS and SUBSCRIBERS. A FETCHER simply requests the current value of some PRESENTITY's PRESENCE INFORMATION from the PRESENCE SERVICE. In contrast, a SUBSCRIBER requests notification from the PRESENCE SERVICE of

(future) changes in some PRESENTITY's PRESENCE INFORMATION. A special kind of FETCHER is one that fetches information on a regular basis. This is called a POLLER.

```
+----------------WATCHER---------------+
|                                      |
|  +----FETCHER---+  +--SUBSCRIBER--+  |
|  |              |  |              |  |
|  | +--POLLER--+ |  |              |  |
|  | |          | |  |              |  |
|  | +----------+ |  |              |  |
|  +--------------+  +--------------+  |
+--------------------------------------+
```

Fig. 2: Varieties of WATCHER

The PRESENCE SERVICE also has WATCHER INFORMATION about WATCHERS and their activities in terms of fetching or subscribing to PRESENCE INFORMATION. The PRESENCE SERVICE may also distribute WATCHER INFORMATION to some WATCHERS, using the same mechanisms that are available for distributing PRESENCE INFORMATION.

Changes to PRESENCE INFORMATION are distributed to SUBSCRIBERS via NOTIFICATIONS. Figures 3a through 3c show the flow of information as a piece of PRESENCE INFORMATION is changed from P1 to P2.


```
     +---------------------------+
     |     PRESENCE SERVICE      |
     |           P1              |
     +---------------------------+

  +------------+       +------------+
  |   P1->P2   |       |     P1     |
  | PRESENTITY |       | SUBSCRIBER |
  +------------+       +------------+
```

Fig. 3a: NOTIFICATION (Step 1)

```
     +---------------------------+
     |     PRESENCE SERVICE      |
     |          P1->P2           |
     +---------------------------+
         ^
         |P2
  +------------+       +------------+
  |     P2     |       |     P1     |
  | PRESENTITY |       | SUBSCRIBER |
  +------------+       +------------+
```

Fig. 3b: NOTIFICATION (Step 2)

```
                  +---------------------------+
                  |      PRESENCE SERVICE     |
                  |             P2            |
                  +---------------------------+
                                        |P2
                                        v
               +-------------+      +-------------+
               |     P2      |      |   P1->P2    |
               | PRESENTITY  |      | SUBSCRIBER  |
               +-------------+      +-------------+

                  Fig. 3c: NOTIFICATION (Step 3)
```

2.2 INSTANT MESSAGE SERVICE

The INSTANT MESSAGE SERVICE also has two distinct sets of "clients": SENDERS and INSTANT INBOXES. A SENDER provides INSTANT MESSAGES to the INSTANT MESSAGE SERVICE for delivery. Each INSTANT MESSAGE is

addressed to a particular INSTANT INBOX ADDRESS, and the INSTANT MESSAGE SERVICE attempts to deliver the message to a corresponding INSTANT INBOX.

```
                 +---------------------------+
                 |  INSTANT MESSAGE SERVICE  |
                 |                           |
                 +---------------------------+
                     ^                 |
                     |                 |
                     |                 v
              +-------------+      +---------------+
              |   SENDER    |      | INSTANT INBOX |
              +-------------+      +---------------+

             Fig. 4: Overview of Instant Message Service
```

2.3 Protocols

A PRESENCE PROTOCOL defines the interaction between PRESENCE SERVICE, PRESENTITIES, and WATCHERS. PRESENCE INFORMATION is carried by the PRESENCE PROTOCOL.

An INSTANT MESSAGE PROTOCOL defines the interaction between INSTANT MESSAGE SERVICE, SENDERS, and INSTANT INBOXES. INSTANT MESSAGES are carried by the INSTANT MESSAGE PROTOCOL.

In terms of this model, we believe that the IMPP working group is planning to develop detailed requirements and specifications for the structure and formats of the PRESENCE PROTOCOL, PRESENCE INFORMATION, INSTANT MESSAGE PROTOCOL, and INSTANT MESSAGES.

2.4 Formats

The model defines the PRESENCE INFORMATION to consist of an arbitrary number of elements, called PRESENCE TUPLES. Each such element consists of a STATUS marker (which might convey information such as online/offline/busy/away/do not disturb), an optional COMMUNICATION ADDRESS, and optional OTHER PRESENCE MARKUP. A COMMUNICATION ADDRESS includes a COMMUNICATION MEANS and a CONTACT ADDRESS. One type of

COMMUNICATION MEANS, and the only one defined by this model, is INSTANT MESSAGE SERVICE. One type of CONTACT ADDRESS, and the only one defined by this model, is INSTANT INBOX ADDRESS. However, other possibilities exist: a COMMUNICATION MEANS might indicate some form of telephony, for example, with the corresponding CONTACT ADDRESS containing a telephone number.


```
+-------------------------------------+
| PRESENCE INFORMATION                |
+-------------------------------------+
 | +--------------------------------+
 =>| PRESENCE TUPLE                 |
 | +--------------------------------+
 |     | +-------------------------+
 |    =>| STATUS                   |
 |     | +-------------------------+
 |     | +-------------------------+
 |    =>| COMMUNICATION ADDRESS    |
 |     | +-------------------------+
 |     |       | +-----------------+
 |     |      =>| CONTACT MEANS    |
 |     |       | +-----------------+
 |     |       | +-----------------+
 |     |      =>| CONTACT ADDRESS  |
 |     |         +-----------------+
 |     | +-------------------------+
 |    =>| OTHER MARKUP             |
 |       +-------------------------+
 | +--------------------------------+
 =>| PRESENCE TUPLE                 |
 | +--------------------------------+
 |     | +-------------------------+
 |    =>| STATUS                   |
 |     | +-------------------------+
 |     | +-------------------------+
 |    =>| COMMUNICATION ADDRESS    |
 |     | +-------------------------+
 |     |       | +-----------------+
 |     |      =>| CONTACT MEANS    |
 |     |       | +-----------------+
 |     |       | +-----------------+
 |     |      =>| CONTACT ADDRESS  |
 |     |         +-----------------+
 |     | +-------------------------+
 |    =>| OTHER MARKUP             |
 |       +-------------------------+
 | +--------------------------------+
 =>| PRESENCE TUPLE                 |
 | +--------------------------------+
 |     ...
```

Fig. 5: The structure of PRESENCE INFORMATION

□

STATUS is further defined by the model to have at least two states that interact with INSTANT MESSAGE delivery -- OPEN, in which INSTANT MESSAGES will be accepted, and CLOSED, in which INSTANT MESSAGES will not be accepted. OPEN and CLOSED may also be applicable to other COMMUNICATION MEANS -- OPEN mapping to some state meaning "available" or "open for business" while CLOSED means "unavailable" or "closed to business." The model allows STATUS to include other values, which may be interpretable by programs or only by persons. The model also allows STATUS to consist of single or multiple values.

2.5 Presence and its effect on Instant Messages

An INSTANT INBOX is a receptacle for INSTANT MESSAGES. Its INSTANT INBOX ADDRESS is the information that can be included in PRESENCE INFORMATION to define how an INSTANT MESSAGE should be delivered to that INSTANT INBOX. As noted above, certain values of the STATUS marker indicate whether INSTANT MESSAGES will be accepted at the INSTANT INBOX. The model does not otherwise constrain the delivery mechanism or format for instant messages. Reasonable people can disagree about whether this omission is a strength or a weakness of this model.

2.6 PRINCIPALS and their agents

This model includes other elements that are useful in characterizing how the protocol and markup work. PRINCIPALS are the people, groups, and/or software in the "real world" outside the system that use the system as a means of coordination and communication. It is entirely outside the model how the real world maps onto PRINCIPALS -- the system of model entities knows only that two distinct PRINCIPALS are distinct, and two identical PRINCIPALS are identical.

A PRINCIPAL interacts with the system via one of several user agents (INBOX USER AGENT; SENDER USER AGENT; PRESENCE USER AGENT; WATCHER USER AGENT). As usual, the different kinds of user agents are split apart in this model even though most implementations will combine at least some of them. A user agent is purely coupling between a PRINCIPAL and some core entity of the system (respectively, INSTANT INBOX; SENDER; PRESENTITY; WATCHER).

□


```
                +---------------------------+
                |     PRESENCE SERVICE      |
                +---------------------------+
                    ^                   |
                    | PRESENCE PROTOCOL |
                    |                   v
```

```
                 +------------+       +------------+
                 | PRESENTITY |       |  WATCHER   |
                 +------------+       +------------+
                       ^                    ^
                       |                    |
                       |                    |
     o          +--------------+      +-------------+       o
    /|\   -->|  PRESENCE UA  |      | WATCHER UA  |<--   /|\
     X          +--------------+      +-------------+       X

 (PRINCIPAL)                                         (PRINCIPAL)

                   Fig. 6: A presence system
```

```
                  +---------------------------+
                  |  INSTANT MESSAGE SERVICE  |
                  +---------------------------+
                       ^                  |
                     IM|  INSTANT MESSAGE |IM
                       |     PROTOCOL     v
                +------------+       +---------------+
                |   SENDER   |       | INSTANT INBOX |
                +------------+       +---------------+
                      ^                     ^
                      |                     |
                      |                     |
     o          +-------------+      +------------------+       o
    /|\   -->|  SENDER UA  |      |  INBOX UA        |<--   /|\
     X          +-------------+      +------------------+       X

 (PRINCIPAL)                                              (PRINCIPAL)

               Fig. 7: An instant messaging system
```


2.7 Examples

A simple example of applying the model is to describe a generic "buddy list" application. These applications typically expose the user's presence to others, and make it possible to see the presence of others. So we could describe a buddy list as the combination of a PRESENCE USER AGENT and WATCHER USER AGENT for a single PRINCIPAL, using a single PRESENTITY and a single SUBSCRIBER.

We could then extend our example to instant messaging and describe a generic "instant messenger" as essentially a buddy list with additional capabilities for sending and receiving instant messages. So an instant messenger would be the combination of a PRESENCE USER AGENT, WATCHER USER AGENT, INBOX USER AGENT, and SENDER USER AGENT for a single PRINCIPAL, using a single PRESENTITY, single SUBSCRIBER, and single INSTANT INBOX, with the PRESENTITY's PRESENCE INFORMATION including an INSTANT INBOX ADDRESS that leads to the INSTANT INBOX.

3. Model

ACCESS RULES: constraints on how a PRESENCE SERVICE makes PRESENCE INFORMATION available to WATCHERS. For each PRESENTITY's PRESENCE INFORMATION, the applicable ACCESS RULES are manipulated by the PRESENCE USER AGENT of a PRINCIPAL that controls the PRESENTITY.

Motivation: We need some way of talking about hiding presence information from people.

CLOSED: a distinguished value of the STATUS marker. In the context of INSTANT MESSAGES, this value means that the associated INSTANT INBOX ADDRESS, if any, corresponds to an INSTANT INBOX that is unable to accept an INSTANT MESSAGE. This value may have an analogous meaning for other COMMUNICATION MEANS, but any such meaning is not defined by this model. Contrast with OPEN.

COMMUNICATION ADDRESS: consists of COMMUNICATION MEANS and CONTACT ADDRESS.

COMMUNICATION MEANS: indicates a method whereby communication can take place. INSTANT MESSAGE SERVICE is one example of a COMMUNICATION MEANS.

CONTACT ADDRESS: a specific point of contact via some COMMUNICATION MEANS. When using an INSTANT MESSAGE SERVICE, the CONTACT ADDRESS is an INSTANT INBOX ADDRESS.

DELIVERY RULES: constraints on how an INSTANT MESSAGE SERVICE delivers received INSTANT MESSAGES to INSTANT INBOXES. For each INSTANT INBOX, the applicable DELIVERY RULES are manipulated by the INBOX USER AGENT of a PRINCIPAL that controls the INSTANT INBOX.

Motivation: We need a way of talking about filtering instant messages.

FETCHER: a form of WATCHER that has asked the PRESENCE SERVICE to for the PRESENCE INFORMATION of one or more PRESENTITIES, but has not asked for a SUBSCRIPTION to be created.

INBOX USER AGENT: means for a PRINCIPAL to manipulate zero or more INSTANT INBOXES controlled by that PRINCIPAL.

Motivation: This is intended to isolate the core functionality of an INSTANT INBOX from how it might appear to be manipulated by a product. This manipulation includes fetching messages, deleting messages, and setting DELIVERY RULES. We deliberately take no position on whether the INBOX USER AGENT, INSTANT INBOX, and INSTANT MESSAGE SERVICE are colocated or distributed across machines.

INSTANT INBOX: receptacle for INSTANT MESSAGES intended to be read by the INSTANT INBOX's PRINCIPAL.

INSTANT INBOX ADDRESS: indicates whether and how the PRESENTITY's

PRINCIPAL can receive an INSTANT MESSAGE in an INSTANT INBOX. The STATUS and INSTANT INBOX ADDRESS information are sufficient to determine whether the PRINCIPAL appears ready to accept the INSTANT MESSAGE.

Motivation: The definition is pretty loose about exactly how any of this works, even leaving open the possibility of reusing parts of the email infrastructure for instant messaging.

INSTANT MESSAGE: an identifiable unit of data, of small size, to be sent to an INSTANT INBOX.

Motivation: We do not define "small" but we seek in this definition to avoid the possibility of transporting an arbitrary-length stream labelled as an "instant message."

INSTANT MESSAGE PROTOCOL: The messages that can be exchanged between a SENDER USER AGENT and an INSTANT MESSAGE SERVICE, or between an INSTANT MESSAGE SERVICE and an INSTANT INBOX.

INSTANT MESSAGE SERVICE: accepts and delivers INSTANT MESSAGES.

-- May require authentication of SENDER USER AGENTS and/or INSTANT INBOXES.

-- May have different authentication requirements for different INSTANT INBOXES, and may also have different authentication requirements for different INSTANT INBOXES controlled by a single PRINCIPAL.

-- May have an internal structure involving multiple SERVERS and/or PROXIES. There may be complex patterns of redirection and/or proxying while retaining logical connectivity to a single INSTANT MESSAGE SERVICE. Note that an INSTANT MESSAGE SERVICE does not require having a distinct SERVER -- the service may be implemented as direct communication between SENDER and INSTANT INBOX.

-- May have an internal structure involving other INSTANT MESSAGE SERVICES, which may be independently accessible in their own right as well as being reachable through the initial INSTANT MESSAGE SERVICE.

NOTIFICATION: a message sent from the PRESENCE SERVICE to a SUBSCRIBER when there is a change in the PRESENCE INFORMATION of some PRESENTITY of interest, as recorded in one or more SUBSCRIPTIONS.

Motivation: We deliberately take no position on what part of the changed information is included in a NOTIFICATION.

OPEN: a distinguished value of the STATUS marker. In the context of INSTANT MESSAGES, this value means that the associated INSTANT INBOX ADDRESS, if any, corresponds to an INSTANT INBOX that is ready to accept an INSTANT MESSAGE. This value may have an

analogous meaning for other COMMUNICATION MEANS, but any such meaning is not defined by this model. Contrast with CLOSED.

OTHER PRESENCE MARKUP: any additional information included in the PRESENCE INFORMATION of a PRESENTITY. The model does not define this further.

POLLER: a FETCHER that requests PRESENCE INFORMATION on a regular basis.

PRESENCE INFORMATION: consists of one or more PRESENCE TUPLES.

PRESENCE PROTOCOL: The messages that can be exchanged between a PRESENTITY and a PRESENCE SERVICE, or a WATCHER and a PRESENCE SERVICE.

PRESENCE SERVICE: accepts, stores, and distributes PRESENCE INFORMATION.

-- May require authentication of PRESENTITIES, and/or WATCHERS.

-- May have different authentication requirements for different PRESENTITIES.

-- May have different authentication requirements for different WATCHERS, and may also have different authentication requirements for different PRESENTITIES being watched by a single WATCHER.

-- May have an internal structure involving multiple SERVERS and/or PROXIES. There may be complex patterns of redirection and/or proxying while retaining logical connectivity to a single PRESENCE SERVICE. Note that a PRESENCE SERVICE does not require having a distinct SERVER -- the service may be implemented as direct communication among PRESENTITY and WATCHERS.

-- May have an internal structure involving other PRESENCE SERVICES, which may be independently accessible in their own right as well as being reachable through the initial PRESENCE SERVICE.

PRESENCE TUPLE: consists of a STATUS, an optional COMMUNICATION ADDRESS, and optional OTHER PRESENCE MARKUP.

PRESENCE USER AGENT: means for a PRINCIPAL to manipulate zero or more PRESENTITIES.

Motivation: This is essentially a "model/view" distinction: the PRESENTITY is the model of the presence being exposed, and is independent of its manifestation in any user interface. In addition, we deliberately take no position on how the PRESENCE USER AGENT, PRESENTITY, and PRESENCE SERVICE are colocated or distributed across machines.

PRESENTITY (presence entity): provides PRESENCE INFORMATION to a PRESENCE SERVICE.

Motivation: We don't like to coin new words, but "presentity" seemed worthwhile so as to have an unambiguous term for the entity of interest to a presence service. Note that the presentity is not (usually) located in the presence service: the presence service only has a recent version of the presentity's presence information. The presentity initiates changes in the presence information to be distributed by the presence service.

PRINCIPAL: human, program, or collection of humans and/or programs that chooses to appear to the PRESENCE SERVICE as a single actor, distinct from all other PRINCIPALS.

Motivation: We need a clear notion of the actors outside the system. "Principal" seems as good a term as any.

PROXY: a SERVER that communicates PRESENCE INFORMATION, INSTANT MESSAGES, SUBSCRIPTIONS and/or NOTIFICATIONS to another SERVER. Sometimes a PROXY acts on behalf of a PRESENTITY, WATCHER, or INSTANT INBOX.

SENDER: source of INSTANT MESSAGES to be delivered by the INSTANT MESSAGE SERVICE.

SENDER USER AGENT: means for a PRINCIPAL to manipulate zero or more SENDERS.

SERVER: an indivisible unit of a PRESENCE SERVICE or INSTANT MESSAGE SERVICE.

SPAM: unwanted INSTANT MESSAGES.

SPOOFING: a PRINCIPAL improperly imitating another PRINCIPAL.

STALKING: using PRESENCE INFORMATION to infer the whereabouts of a PRINCIPAL, especially for malicious or illegal purposes.

STATUS: a distinguished part of the PRESENCE INFORMATION of a PRESENTITY. STATUS has at least the mutually-exclusive values OPEN and CLOSED, which have meaning for the acceptance of INSTANT MESSAGES, and may have meaning for other COMMUNICATION MEANS. There may be other values of STATUS that do not imply anything about INSTANT MESSAGE acceptance. These other values of STATUS may be combined with OPEN and CLOSED or they may be mutually-exclusive with those values.

Some implementations may combine STATUS with other entities. For example, an implementation might make an INSTANT INBOX ADDRESS visible only when the INSTANT INBOX can accept an INSTANT MESSAGE. Then, the existence of an INSTANT INBOX ADDRESS implies OPEN, while its absence implies CLOSED.

SUBSCRIBER: a form of WATCHER that has asked the PRESENCE SERVICE to notify it immediately of changes in the PRESENCE INFORMATION of one or more PRESENTITIES.

SUBSCRIPTION: the information kept by the PRESENCE SERVICE about a SUBSCRIBER's request to be notified of changes in the PRESENCE INFORMATION of one or more PRESENTITIES.

VISIBILITY RULES: constraints on how a PRESENCE SERVICE makes WATCHER INFORMATION available to WATCHERS. For each WATCHER's WATCHER INFORMATION, the applicable VISIBILITY RULES are manipulated by the WATCHER USER AGENT of a PRINCIPAL that controls the WATCHER.

Motivation: We need a way of talking about hiding watcher information from people.

WATCHER: requests PRESENCE INFORMATION about a PRESENTITY, or WATCHER INFORMATION about a WATCHER, from the PRESENCE SERVICE. Special types of WATCHER are FETCHER, POLLER, and SUBSCRIBER.

WATCHER INFORMATION: information about WATCHERS that have received PRESENCE INFORMATION about a particular PRESENTITY within a particular recent span of time. WATCHER INFORMATION is maintained by the PRESENCE SERVICE, which may choose to present it in the same form as PRESENCE INFORMATION; that is, the service may choose to make WATCHERS look like a special form of PRESENTITY.

Motivation: If a PRESENTITY wants to know who knows about it, it is not enough to examine only information about SUBSCRIPTIONS. A WATCHER might repeatedly fetch information without ever subscribing. Alternately, a WATCHER might repeatedly subscribe, then cancel the SUBSCRIPTION. Such WATCHERS should be visible to the PRESENTITY if the PRESENCE SERVICE offers WATCHER INFORMATION, but will not be appropriately visible if the WATCHER INFORMATION includes only SUBSCRIPTIONS.

WATCHER USER AGENT: means for a PRINCIPAL to manipulate zero or more WATCHERS controlled by that PRINCIPAL.

Motivation: As with PRESENCE USER AGENT and PRESENTITY, the distinction here is intended to isolate the core functionality of a WATCHER from how it might appear to be manipulated by a product. As pre